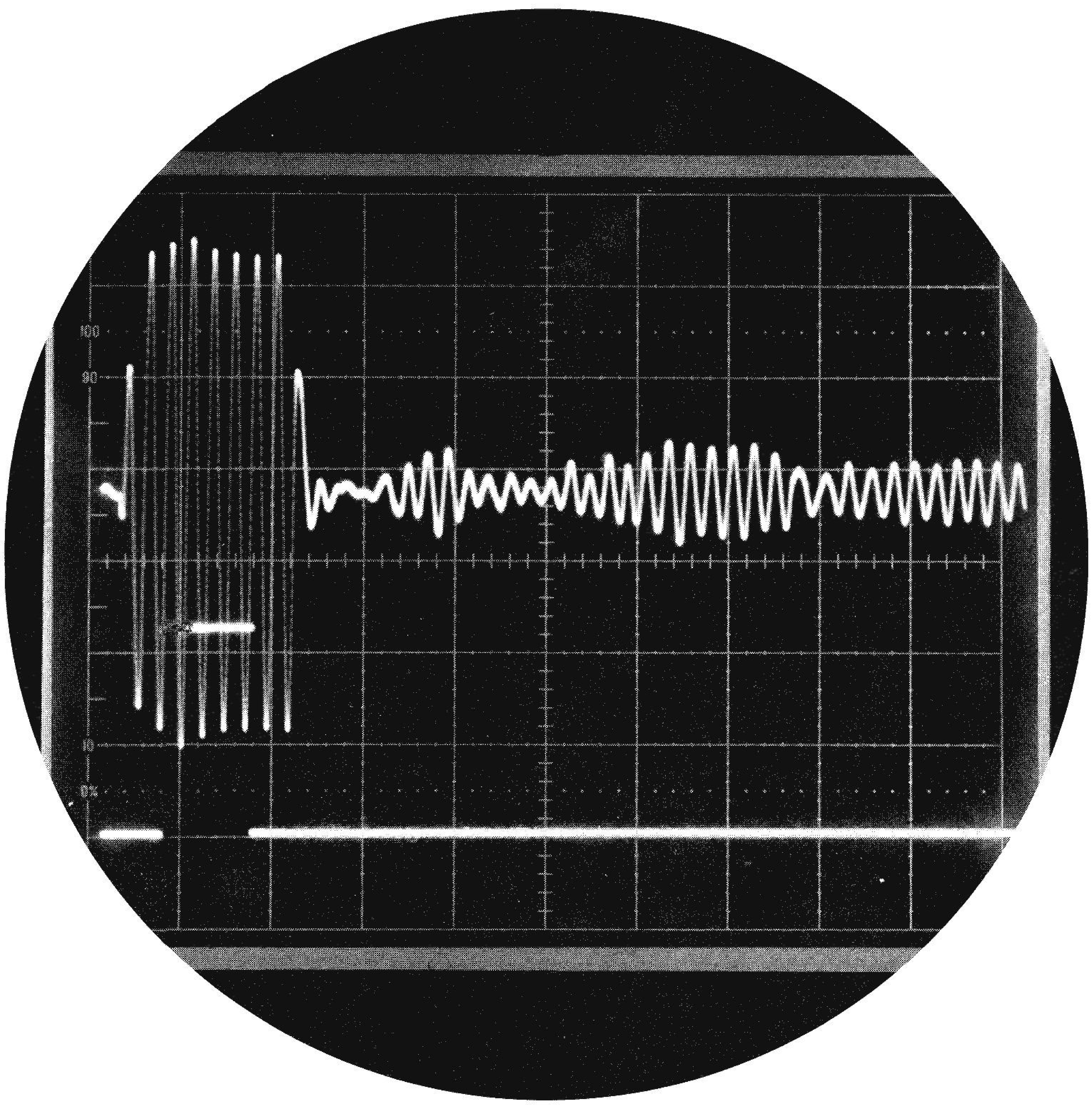
Brüel & Kjær

application notes

Electro Acoustic free field measurements in ordinary rooms



— using gating techniques

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Electroacoustic free-field measurements in ordinary rooms — using gating techniques

by *Henning Møller* and *Carsten Thomsen,* Brüel & Kjær

Introduction

By the use of gating techniques, many types of acoustic measurement normally confined to anechoic rooms, can now be made in ordinary reflective rooms. Of primary importance are the measurements of loudspeaker frequency response and directional characteristics. However, the technique is also applicable to other areas such as measurement of distortion, early reflections, absorption, and phase response. These measurements are made possible by means of a Gating System which provides the necessary tone burst test signal and measures the peak amplitude of the received signal, which is gated by an adjustable time window to eliminate the influence of reflections and loudspeaker transient distortion.

Basic Principles

The Gating Concept

When a tone burst is applied to a loudspeaker, the acoustic waveform received at the position of the measuring microphone often bears little resemblence to the original electrical signal (Fig.1). This is due to a number of phenomena: (1) overshoot of the loudspeaker on the initial part of the burst (2) internal reflections in the loudspeaker cabinet which most often distort the first part of the burst (3) overhang of the loudspeaker at the end of the burst (4) reflections from the walls, floor, and ceiling of the room, which usually arrive after the end of the burst. All of this information is undesirable when wanting to determine the free-field sine wave response of the loudspeaker, although it may be useful in describing other aspects of the loudspeaker's performance. However, somewhere in the received tone burst, a steady-state sine wave can be found whose amplitude equals the free-field response of the loudspeaker. This

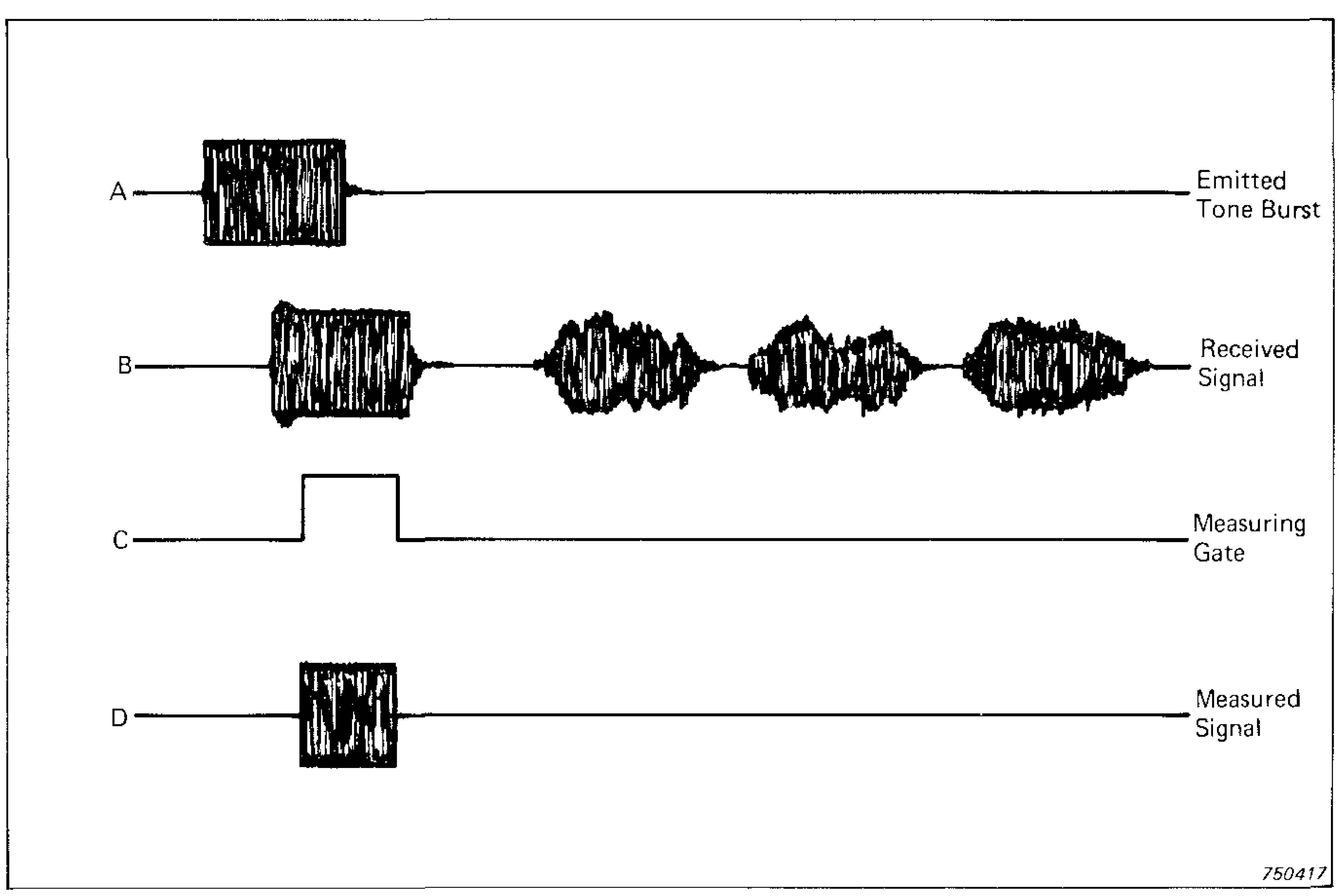

Fig.1. Principle of the Gating System

steady-state information can be measured by the use of a time window (gate) which is adjusted to select only the desired section of the burst (Fig.2).

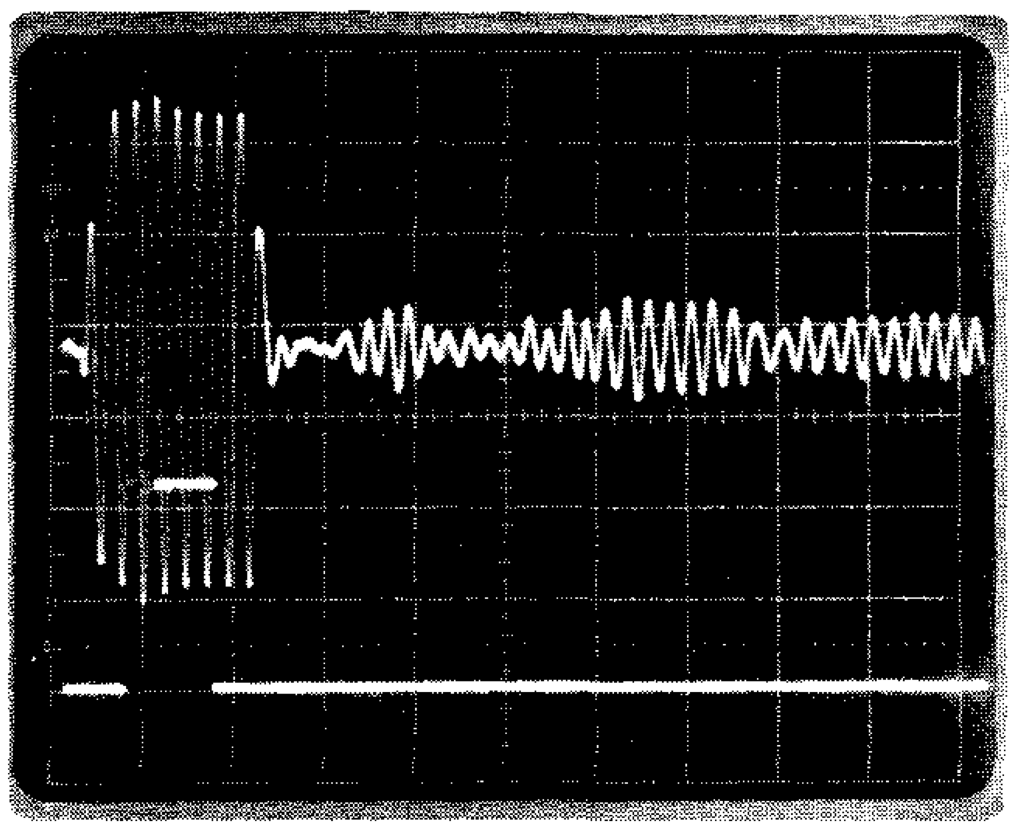

Fig.2. The gate is adjusted to include only the steady state portion of the signal

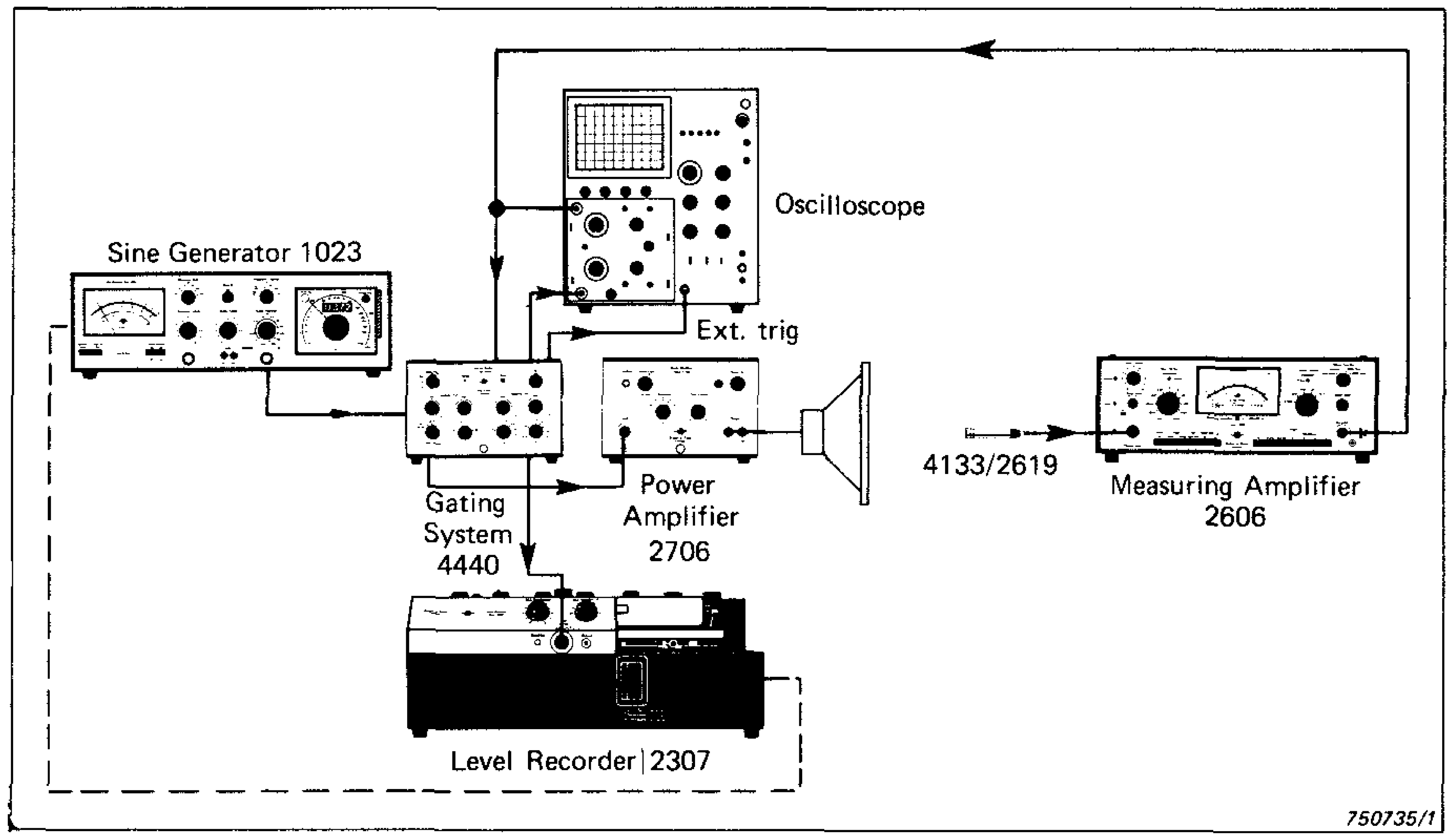

Fig.3. Instrument set-up using the Gating System

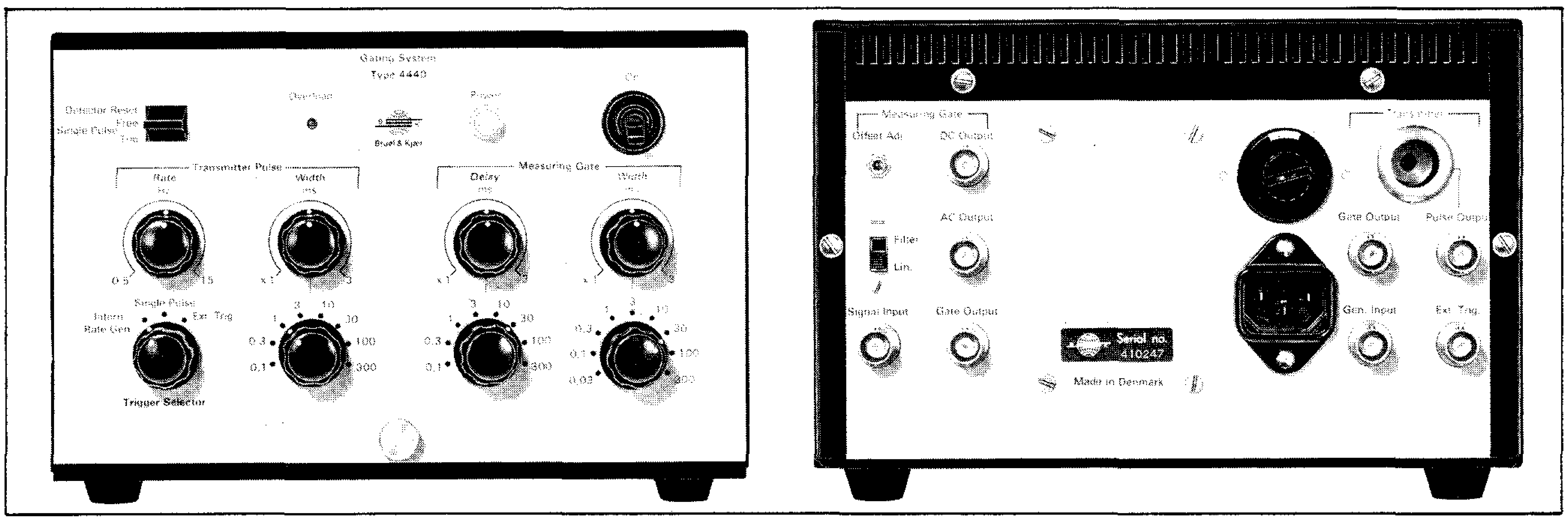

Fig.4. Front and rear view of the Gating System

The measurement of the loudspeaker's amplitude response using this method will be accurate provided that the loudspeaker has fully responded to the input signal as can be seen on the oscilloscope. If the signal has settled to a steady value, the use of a longer tone burst will give no more information. Hence this value will equal the value obtained with a continuous sine wave, which is simply an extremely long tone burst.

Instrumentation

The system for gated measurements builds on the same basic instruments as used for traditional frequency response measurements: a sine generator, measuring microphone, measuring amplifier, and level recorder. To permit gated measurements, the Gating System and a two channel oscilloscope must be added to this basic package (Fig.3).

The Gating System Type 4440 (Fig.4) consists of both a transmitting and receiving section. The sine wave from the Sine Generator 1023 is fed into the transmitting section where it is converted to a tone burst. The duration of the burst is adjustable from 0,1 ms to 1 s and its repetition rate may be varied from 0,5 Hz to 15 Hz, or the burst may be triggered externally. The tone burst is generated so that it always begins and ends on a positive-going zero crossing (Fig.5). This minimizes the generation of other frequency components.

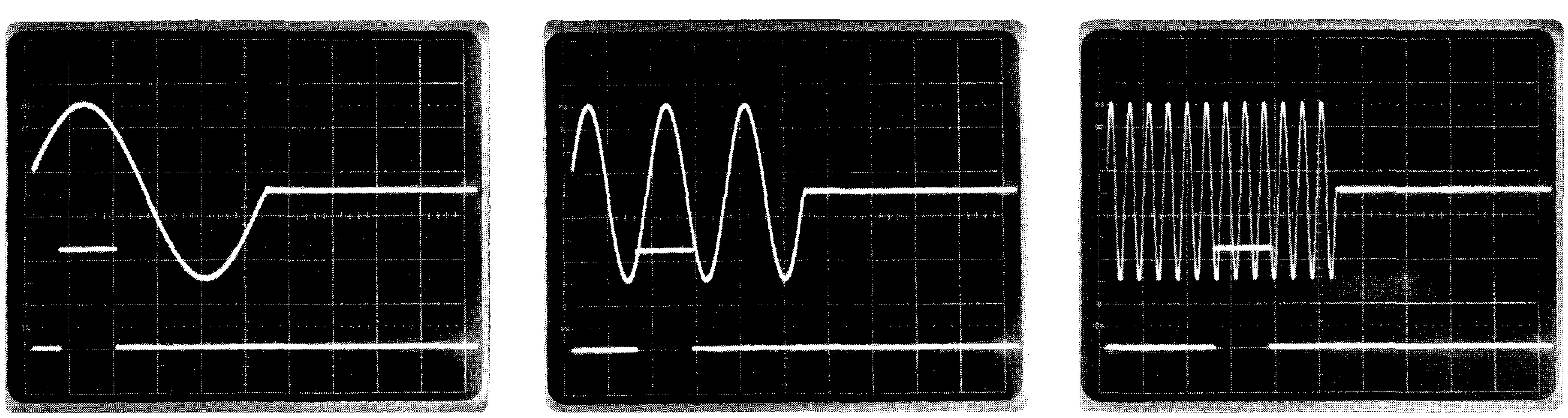

Fig.5. The tone burst always begins and ends with a positive-going zero crossing

In the receiving section, a measuring gate is adjusted to select only the desired portion of the received signal. The width of this measuring gate is adjustable over a wide range, from 30 μs to 1 s. In addition, this "time window" may be delayed from 0,1 ms to 1 s to compensate for the propagation delay of sound in air and to permit selection of the desired section of the burst.

The width and delay of the measuring gate are adjusted with the aid of a two channel oscilloscope. A positive peak detector gives the maximum peak value of the received signal within the gate and holds that value until the next tone burst. The hold circuit is than reset and holds the new peak value of the next burst. This value is fed as a DC voltage to a Level Recorder, which records the automatically swept frequency response. The peak detector operates linearly over a dynamic range of at least 50 dB.

Selection of Microphone Distance and Pulse Length

While the response time of the loudspeaker places a restriction on the minimum pulse length, the maximum pulse length will also be limited to the time it takes for the first reflection to return to the measuring microphone. This, of course, depends on the room dimensions and the positioning of loudspeaker and microphone. It is interesting to note that the lower frequency limits obtained are approximately the same as for anechoic rooms of the same dimensions. In practice, however, the gating system may be used to achieve low frequency performance as good as or better than any anechoic room, provided that a large enough room (such as a warehouse) is available.

First of all, the distance (d) between loudspeaker and microphone should be selected. Ideally, the microphone must be placed in the far field of the loudspeaker, that is, at least one wavelength from the speaker at the lowest frequency ($f_{min.}$)

$$d = \frac{344}{f_{min.}} \qquad (1)$$

The size of the loudspeaker must also be considered in determining the far field. The microphone should be placed at a distance at least equal to the largest dimension of the loudspeaker. Unfortunately, due to practical restrictions on room size, these criteria are often ignored, thus leading to nonreproducible measurements. Certain standards, of course, also call for fixed distances.

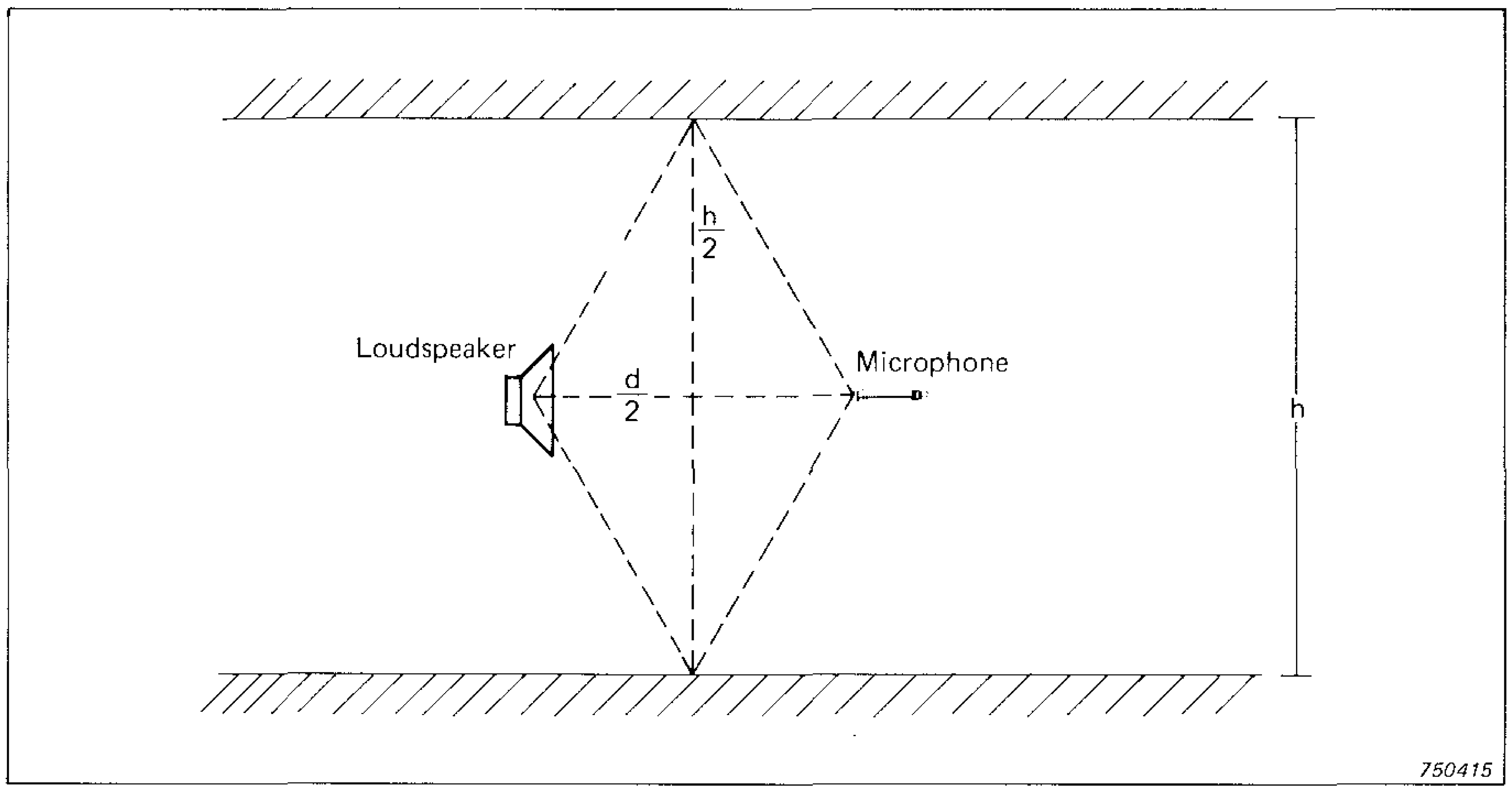

Fig. 6. The microphone/loudspeaker combination should be centered in the room

Optimum Placement

In practice, the microphone and loudspeaker should be placed along the long axis of the room and should be centered between the walls, and the ceiling and floor. (Fig. 6). With this set-up, the optimum distance can be calculated (See Appendix) which fulfils the one wavelength criterion and permits the maximum tone burst length before the first reflection. If the room's smallest dimension is h, then the optimum distance (d) between loudspeaker and microphone is:

$$d = 0{,}58\,h \qquad (2)$$

This is valid when the length of the axis of the room along which the microphone and loudspeaker are placed is at least 15% greater than h. If this is not the case, see Appendix.

The corresponding minimum frequency is

$$f_{min.} = \frac{595}{h} \qquad (3)$$

and the maximum pulse length (t) is $1/f_{min.}$ which at the minimum frequency includes one period of that frequency.

Table 1 shows some practical values for minimum room dimensions of 2 to 5 meters.

h >	2 m	3 m	4 m	5 m
d	1,15 m	1,73 m	2,31 m	2,88 m
$f_{min.}$	298 Hz	199 Hz	149 Hz	119 Hz
t	3,3 ms	5,0 ms	6,7 ms	8,4 ms

Table 1. Optimum microphone/loudspeaker distances (d), lower limiting frequencies ($f_{min.}$) and pulse lengths (t) for given room heights (h)

Pulse Length related to a fixed distance

A fixed, possibly arbitrary distance between transducers, such as 1 m, is often chosen even though it ignores near-field considerations at low frequencies.

In this case the maximum tone burst length is

$$t_{max.} = \frac{M - d}{c} \qquad (4)$$

where M is the shortest reflection path, which for a rectangular room is the smaller of L (length) or $\sqrt{h^2 + d^2}$, and c is the speed of sound. Of course, the minimum frequency again is $1/t_{max.}$

Equation (4) may also be used if there is some reflecting object present other than the room's boundaries, such as a cabinet, or perhaps the measuring instrumentation itself. In this case, the shortest reflection path from the loudspeaker to the reflection object and back to the microphone can simply be measured and substituted as M in Equation (4).

Complete derivations of these considerations are given in the Appendix.

A Practical Measurement Procedure

A suggested measuring procedure is given as follows:

1. Place the loudspeaker and measuring microphone along the longest axis of the room. Center the microphone/loudspeaker combination with respect to all three axes of the room. Choose a suitable distance between the transducers as indicated in Section "Optimum Placement".

2. Connect the instrumentation as indicated in the 4440 Gating System Instruction Manual. Note that there are two GATE OUTPUTs on the rear panel of the 4440, one for the transmitting and one for the measuring section. The transmitting section GATE OUTPUT must be connected to the Ext. Trigger input of the oscilloscope. The measuring section GATE OUTPUT connects to the Channel 2 input of the scope.

3. The received tone burst can be displayed on Channel 1 of the oscilloscope. This signal is available at the output of the Measuring Amplifier. (The signal can also be taken from the AC OUTPUT of the Gating System but must then be inverted 180° to display it properly on the oscilloscope.)

4. Begin with a relatively short tone burst (about 3 ms at about 1 kHz) and observe the received waveform on the oscilloscope. Note the point of the first reflections and increase the duration of the tone burst. If the tone burst is too long, the received signal will appear as in Fig.7, where the direct and reflected signals overlap. This should correspond with the calculated values from section "Selection of Microphone Distance and Pulse Length".

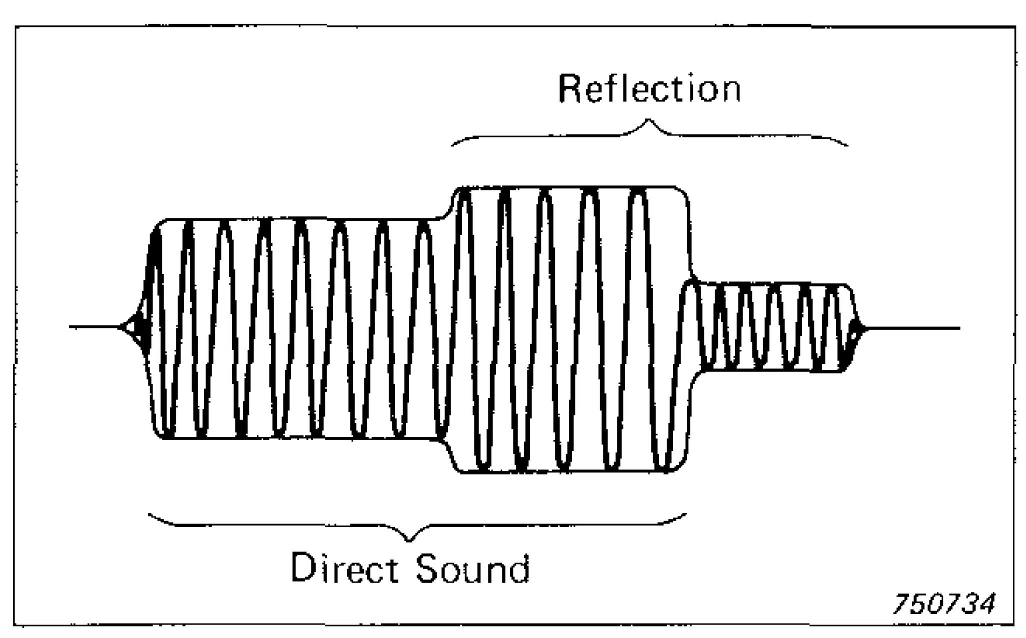

Fig.7. Too long a burst results in overlap of the direct and reflected sound

5. Use a relatively low Repetition Rate and note the distance (in time), along the X-axis of the oscilloscope, that it takes the amplitude of the reflections to drop at least a factor of 10. The Repetition Rate must be adjusted so that the reflections have decayed at least 20 dB before a new burst is transmitted. The more reverberant the room, the slower the required Repetition Rate.

6. Decrease the frequency of the oscillator until only one period is observed on the oscilloscope. At this frequency, adjust the width and delay of the measuring gate to measure the peak value of the second half of the period. If this peak is negative-going, the loudspeaker leads must be reversed to change the polarity. This is necessary because the peak detector of the Gating System Type 4440 only responds to positive peaks. In addition, measuring on the second half of the single period will give a more accurate result since the system has had more time to respond and hence the waveform is less distorted.

   Now sweep the generator through the higher frequencies and if necessary trim the adjustment of the measuring gate so that it only measures the steady-state portion near the end of the tone burst.

7. The rate at which the sweep can be made depends on the tone burst Repetition Rate, and the desired resolution at the maximum frequency of interest. It is important to note that the frequency response measurements obtained using the Gating System are made at discrete frequencies since the gate is only open a short time during each repetition period. Hence details of the curve may be lost if too high a sweep rate is used. Sweep rates given by Level Recorder Paper Speeds of 1 or 3 mm/s are typical. The exact relationship that gives the Paper Speed (P) is:

$$P \leqslant \frac{22\,B\,R}{f_{max.}} \qquad (5)$$

where B is the frequency resolution (in Hz) at the maximum frequency $f_{max.}$ and R is the Repetition Rate. This formula is derived in the Appendix.

Practical Examples

On Axis Response

Ideally, a measurement in an ordinary room using the Gating System should yield the same results as a measurement in an anechoic chamber. A practical comparison is shown in Figs.8 and 9. It will be noted that above 200 Hz only the details of the two curves differ. The difference below 200 Hz is due to the low frequency limits already discussed.

The difference in detail between the two curves is due to the difficulty in exactly duplicating the microphone/loudspeaker position and distance in the two different rooms. Experience shows that even a small angular displacement of the microphone in an anechoic room will yield different results.

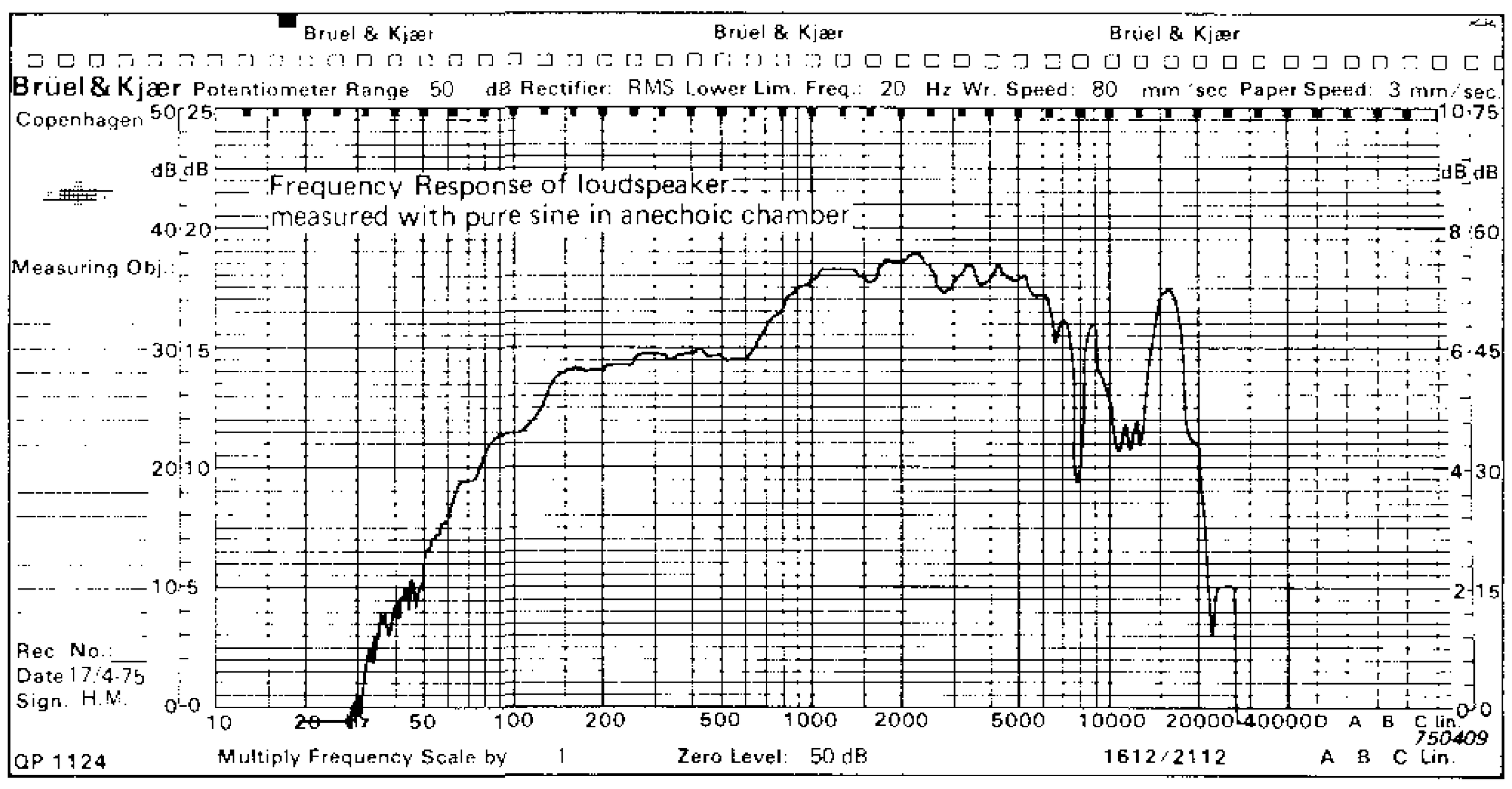

Fig.8. Frequency response of loudspeaker measured with pure sine in anechoic chamber

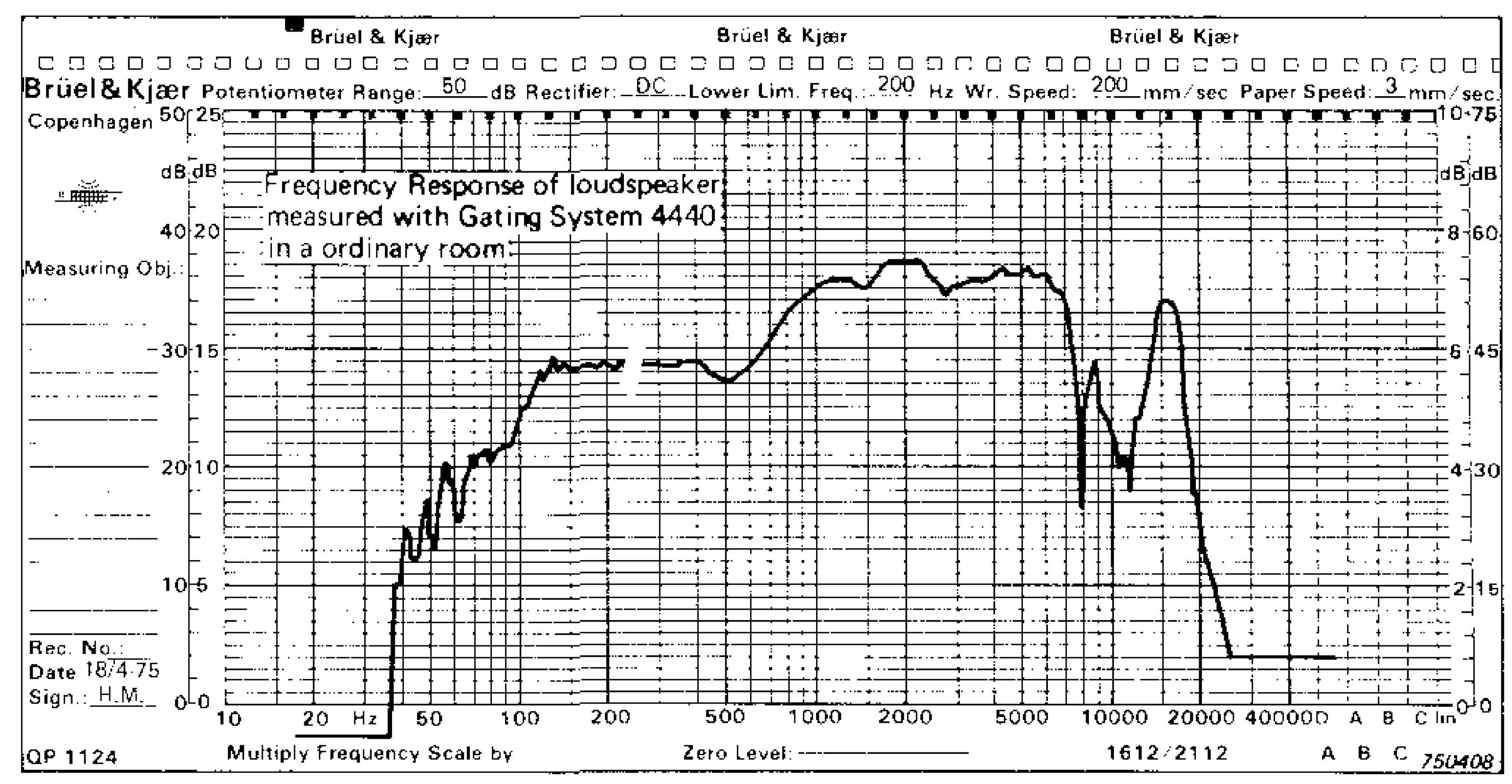

Fig.9. Frequency response of the same loudspeaker measured with the Gating System

To verify this, measurements were made in an anechoic room both with (Fig.10) and without (Fig.11) the Gating System. Thus the microphone position was identical in both measurements and the only source of error will be the influence of the Gating System. From these curves it can be seen that this error is negligible above the lower limiting frequency of about 300 Hz.

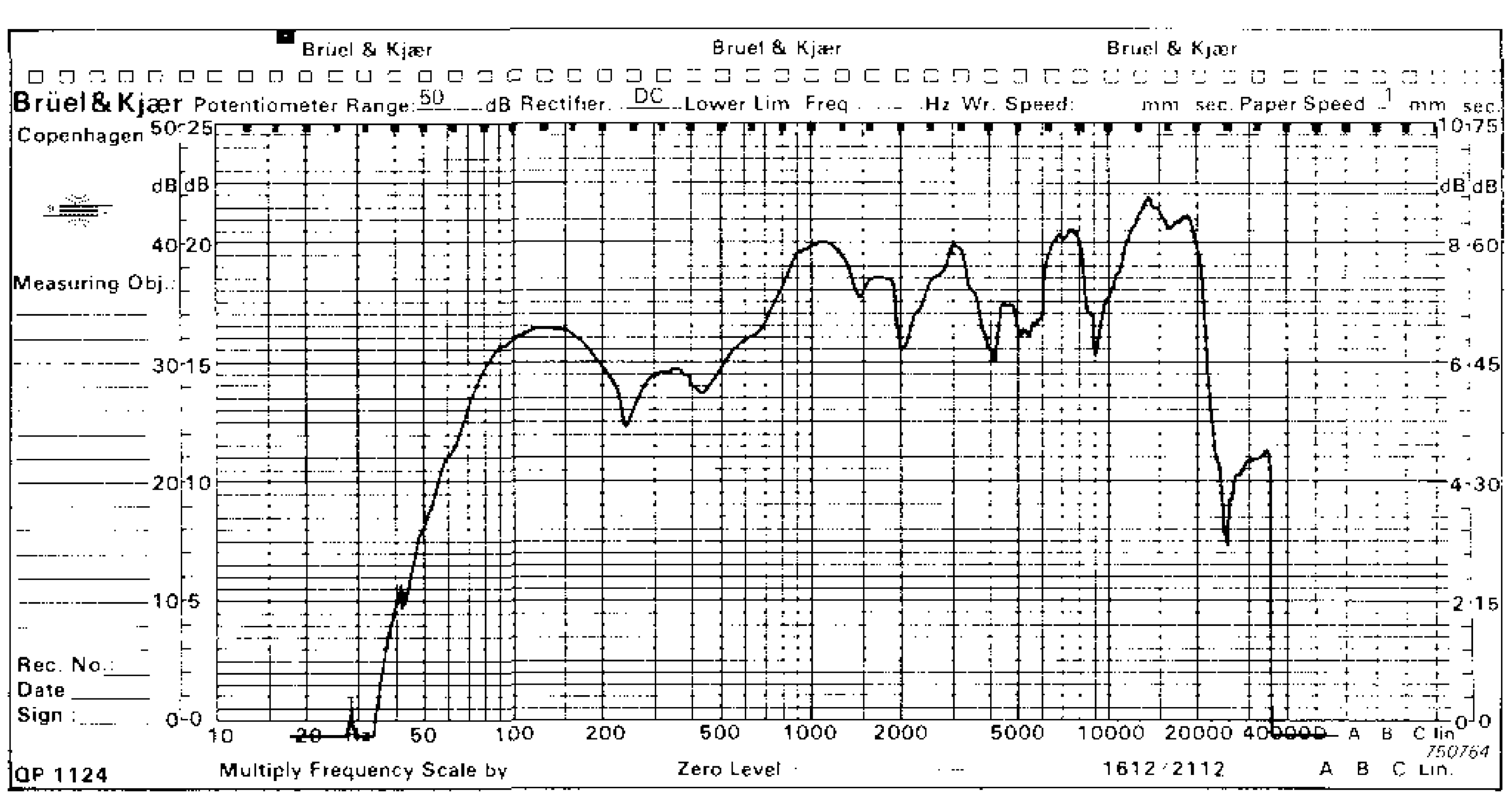

Fig.10. Frequency response of the loudspeaker in Fig.11 measured with Gating System in anechoic room. The difference between these two curves indicates the influence of the Gating System

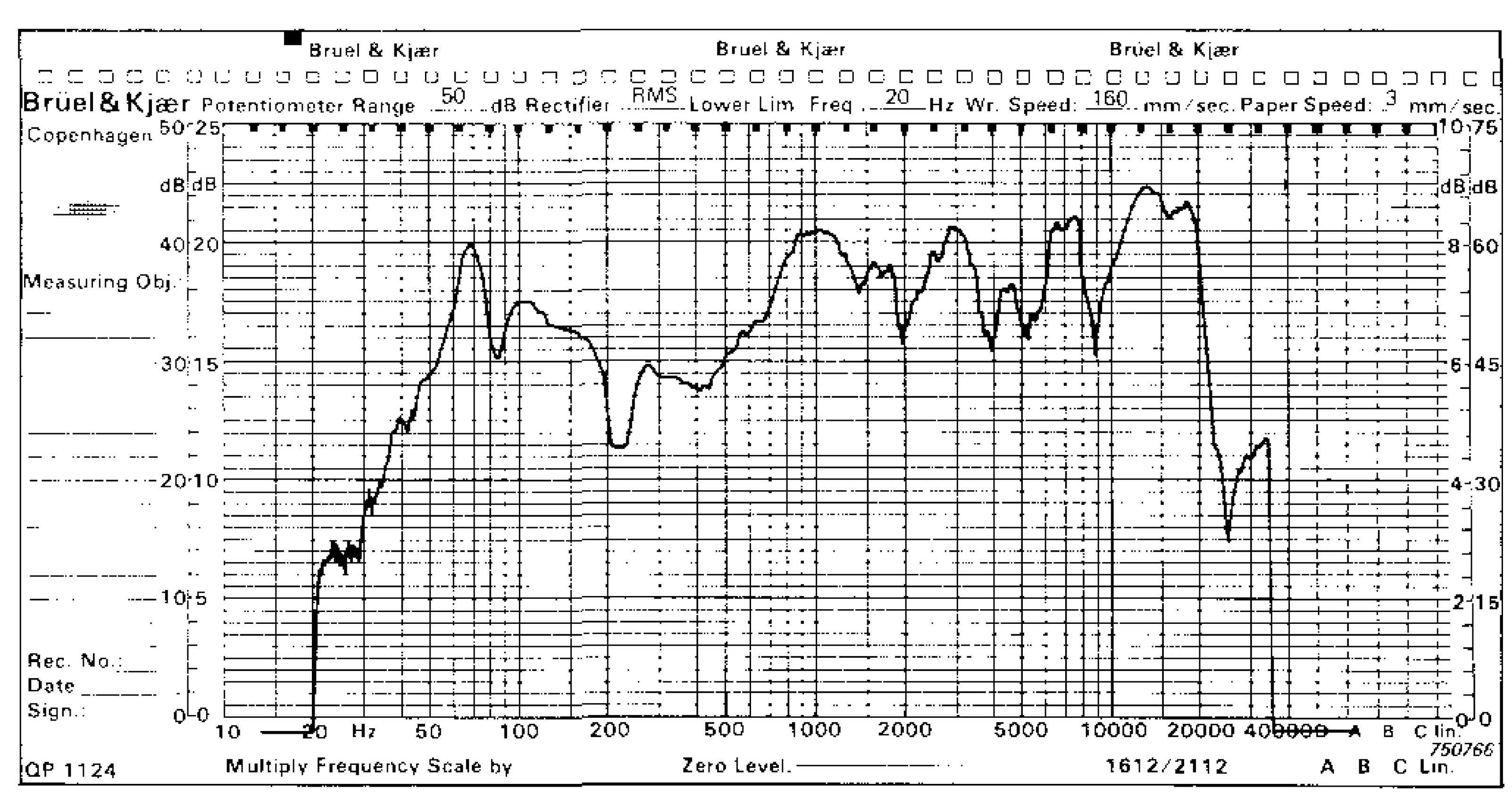

Fig.11. Frequency response of a loudspeaker (not the same as in Figs.8 and 9) using pure sine in anechoic room

The usefulness of the Gating System is seen by comparison to Fig.12. Here a frequency response measurement was made in a reflective room using a continuous sine signal.

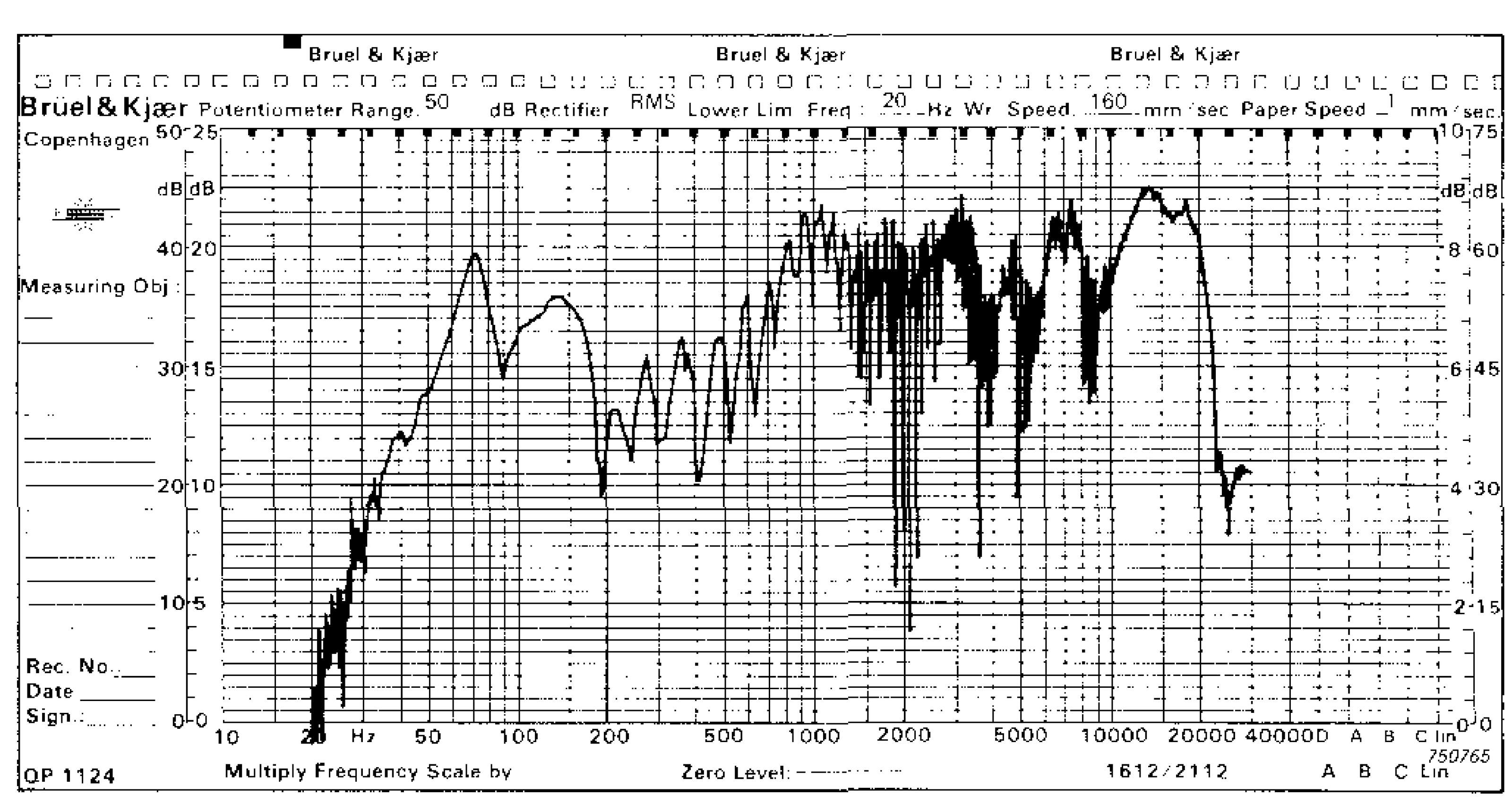

Fig.12. Frequency response of loudspeaker measured in a room with reflecting surfaces

Directional Characteristics

Figures 13 to 16 show the influence of the Gating System on the measurement of the directional characteristics of a loudspeaker. Here we see a good correlation between the two curves. However, it is seen that at the higher frequencies the Gating System levels out some of the very sharp notches. One reason for this can best be explained by describing the cause of these sharp dips. The sound arriving at a given point to the side or rear of a loudspeaker travels by more than one path due to reflection and diffraction phenomena in the loudspeaker. Since the path lengths will differ, the sound signals will reinforce or cancel each other depending on their relative phase. This results in the lobes with the deep notches seen in directional characteristics measured with pure sine. (It is the same kind of standing wave phenomena that causes the sharp dips in Fig.12.)

With the relatively short tone bursts used with the gating system, all the standing wave phenomena do not have time to build-up and hence the sharp dips are not seen. It can be debated which of the two curves is more correct, but this is probably fruitless since the exact shape of the directional characteristic to the rear is of no pratical significance and the numerous lobes and dips will be obliterated by the standing wave patterns of the room in which the speaker is used.

A second reason for the difference in the curves lies in the nature of the tone bursts themselves as described in the next section.

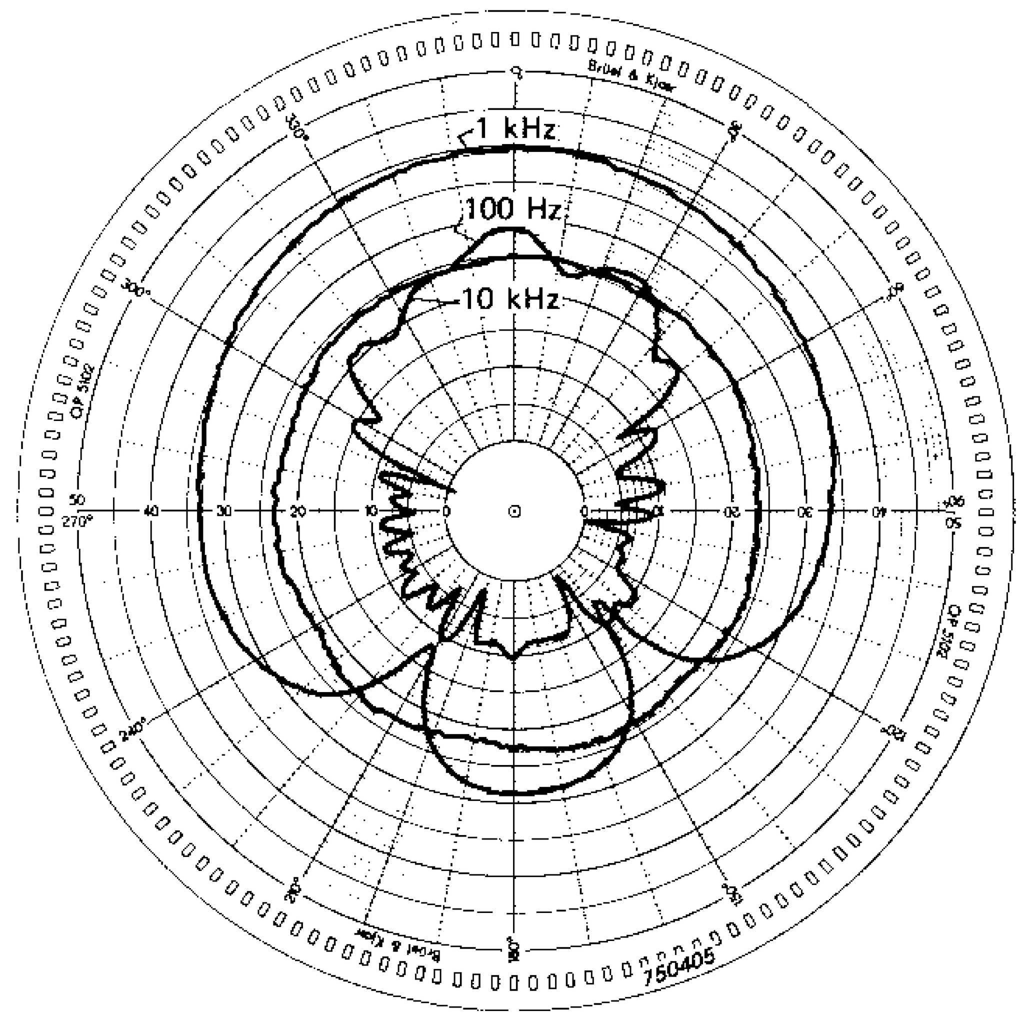

Fig.13. **Directional characteristics of loudspeaker measured with pure sine in anechoic chamber**

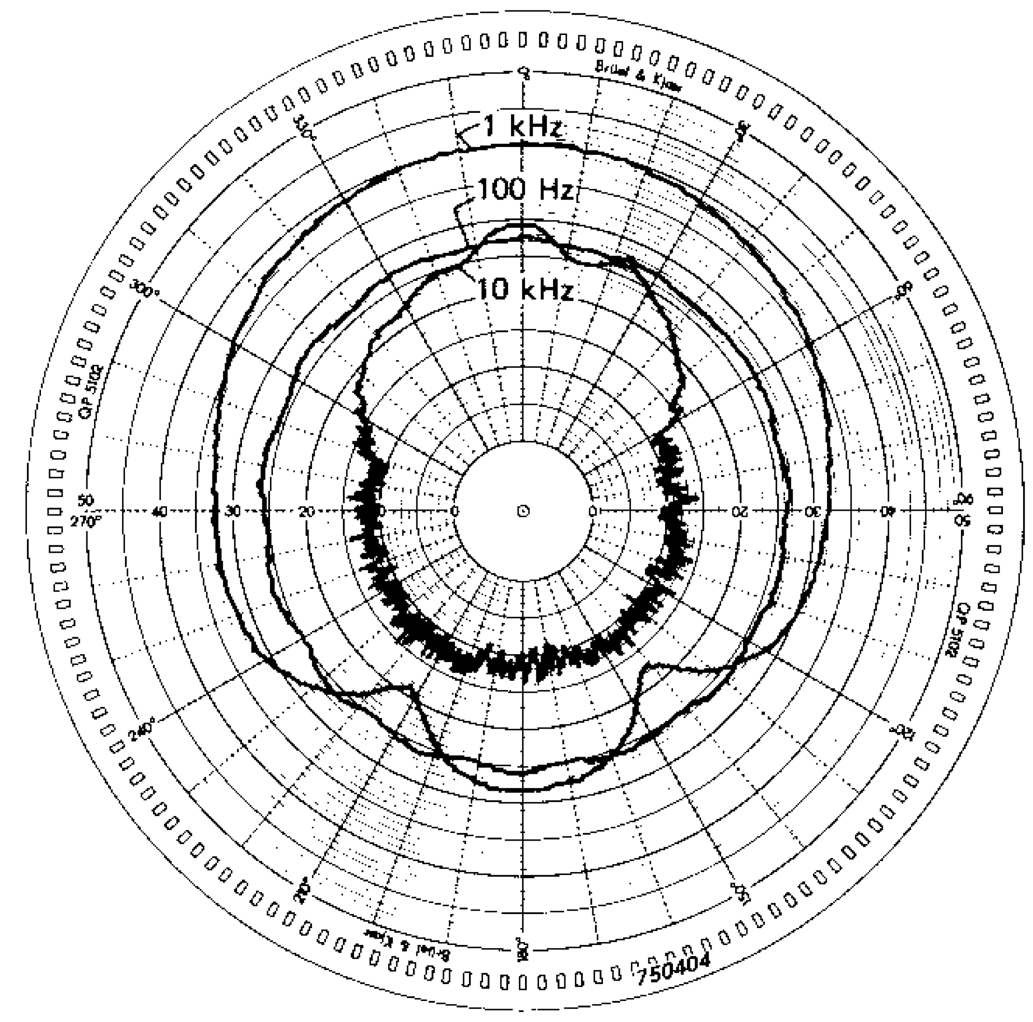

Fig.14. **Directional characteristics of the same loudspeaker measured with the Gating System in an ordinary room. Note the 10 kHz curve shows the background noise level of the room**

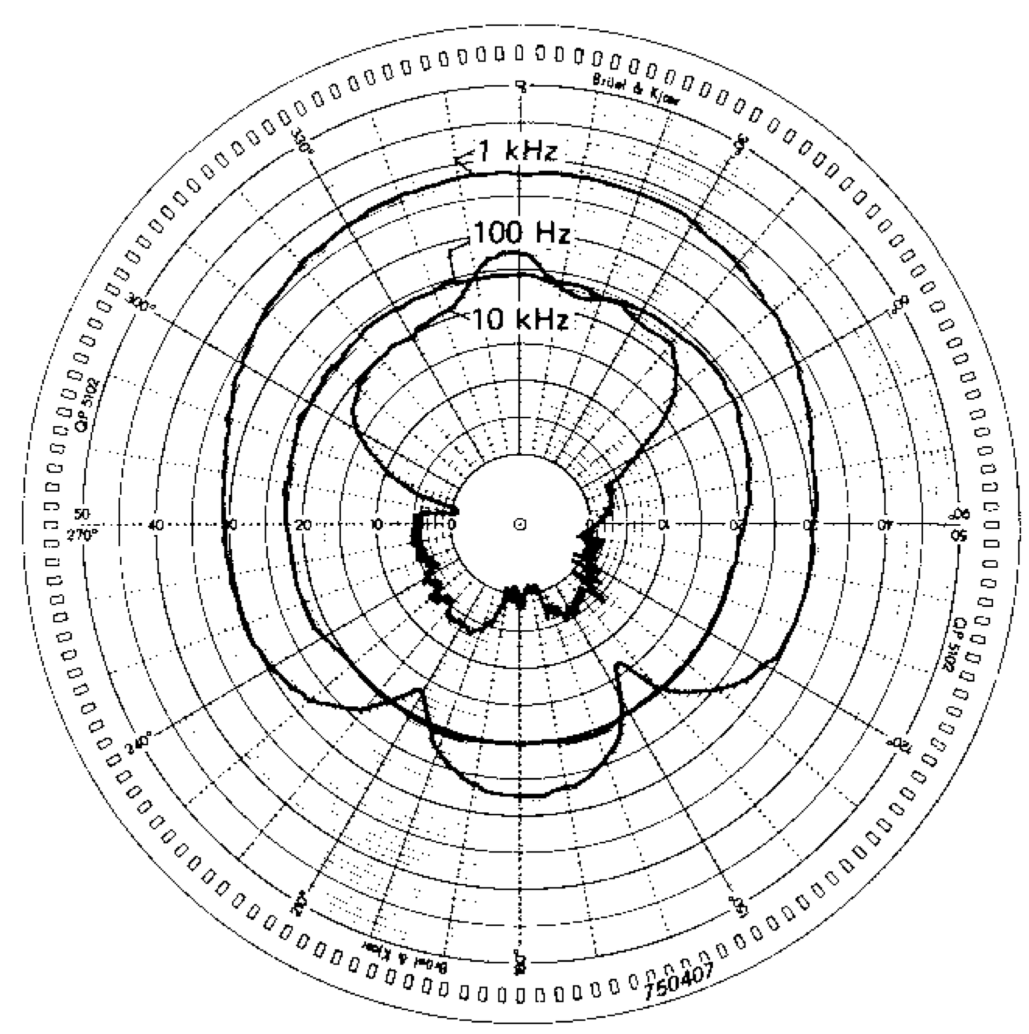

Fig.15. **Directional characteristics of the same loudspeaker measured with the Gating System in an anechoic chamber. The sharp dips and peaks are not as pronounced as for pure sine**

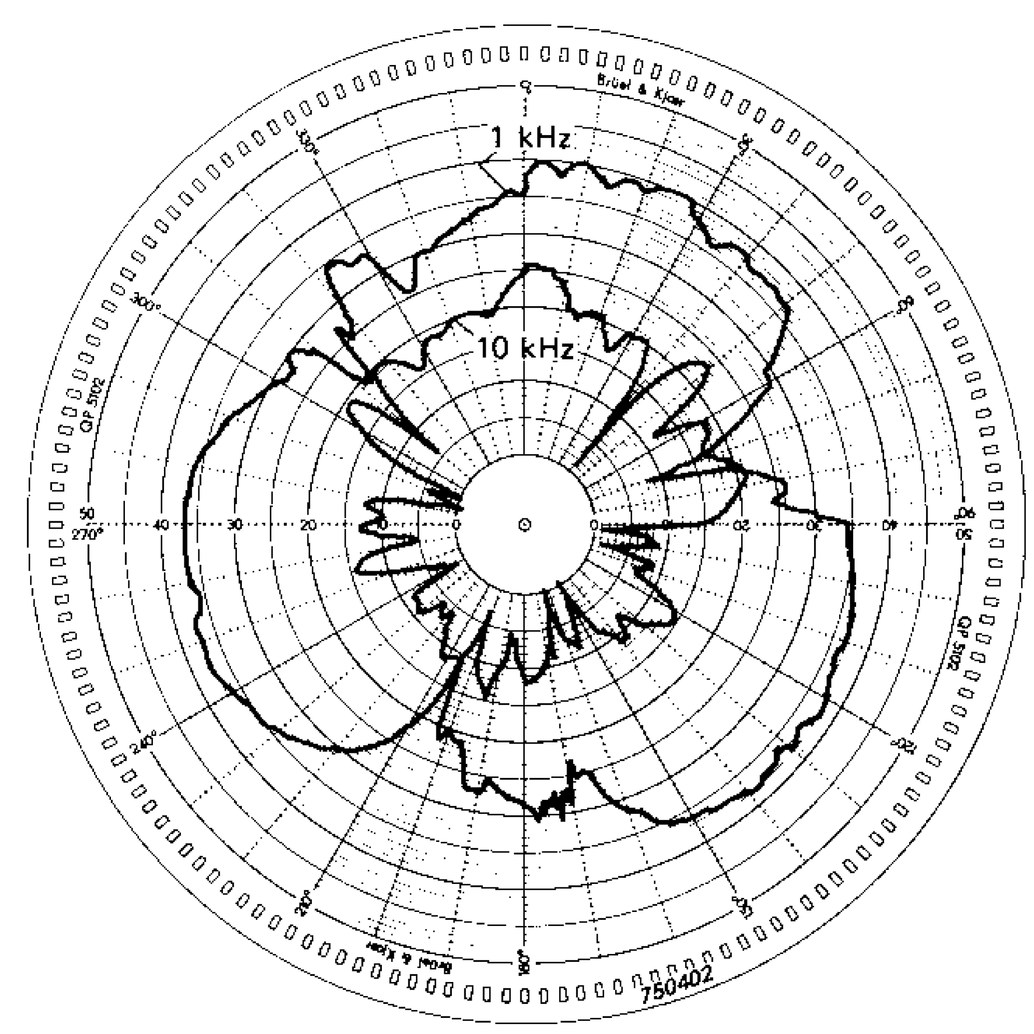

Fig.16. **Directional characteristics of the same loudspeaker measured with pure tones in ordinary room**

Tone Burst Characteristics

A tone burst contains not only the frequency of the sine wave contained in the burst but also a band of frequencies centered around the sine wave frequency. These frequencies arise due to the square wave by which the sine signal is gated. These components can be heard as a "click" at the beginning and end of the tone burst. Physically, these frequency components and the click arise at the abrupt transition from a flat line to sine wave at the beginning and end of the burst.

The frequency spectra of various tone bursts in Fig.17 give us a clearer view of the these phenomena. The spectra are recorded on a Real-Time Narrow Band Analyzer (Type 3348) and are displayed on a linear frequency scale from 0 to 2 kHz. The frequency of the gated sine wave in the burst is 1 kHz and occurs in the middle of the main lobe in the middle of the screen. The width of this lobe is 2/t where t is the duration of the tone burst. Thus a shorter burst gives a broader frequency spectrum. In the limiting case of an infinitely short burst, (Dirac unit impulse) the frequency spectrum will be infinitely wide, i. e. include all frequencies. This case is approached by the 1 ms burst at the top of Fig.17. At the bottom of the figure we see that the 100 ms burst approaches the single line spectrum of a continuous sine wave.

Of particular interest is the —3 dB bandwidth of the center lobe which equals 1/t. Thus a tone burst can be regarded to have a spectrum similar to narrow band noise which is often used as a loudspeaker test signal. (However, the tone burst has numerous side lobes (1/t wide) which the narrow band noise does not have.)

This raises the obvious question: "How can a tone burst with a relatively broad spectrum give as good frequency resolution as a continuous tone which has an infinitely narrow spectrum?" The answer, which we already have hinted at, is hopefully equally obvious. The beginning of the tone burst excites the loudspeaker with a broad spectrum. After a given response time to this "shock" the loudspeaker will settle to the steady-state value of the sine wave. Since we measure only this value, we eliminate the influence of the other frequency components caused by the beginning and end of the tone burst. Thus the frequency resolution of the tone burst can equal that of a continuous sine wave.

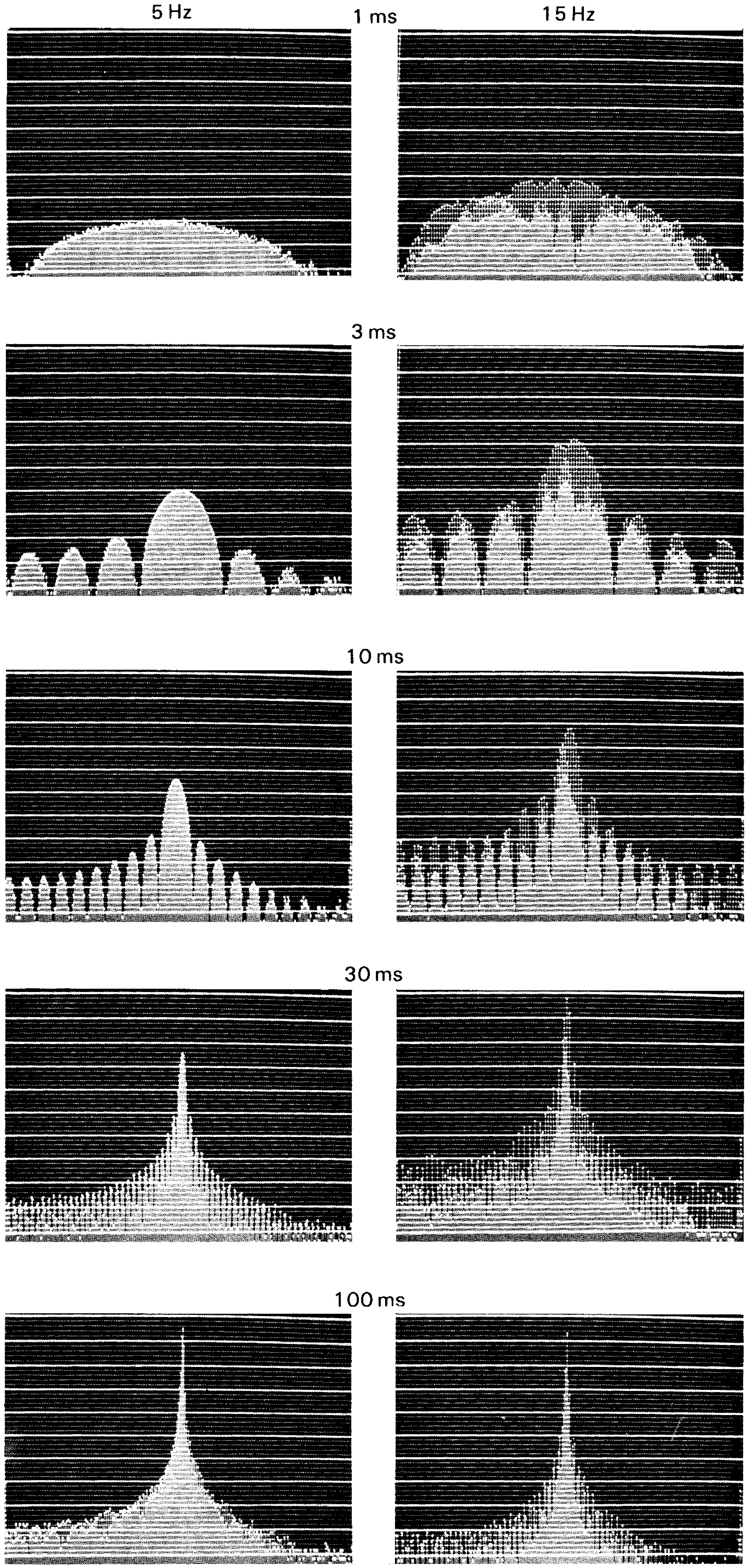

Fig.17. Frequency spectra of tone burts with Repetition Rates of 5 Hz and 15 Hz. Burst width varies from 1 ms to 100 ms. Center frequency is 1 kHz. Displayed on linear scale from 0 to 2 kHz with a 50 dB vertical scale

It is extremely important to emphasize the criterion that the loudspeaker must have fully responded and settled to the steady-state value. If for some reason, the response time of the loudspeaker is longer than the duration of the tone burst, the accuracy of the measurement will be decreased. One cause of the error will be that one does not know which section of the signal to select, since it has not settled. Generally, of course, the section closest to the end of the burst should be most accurate. Since the measurement will not be on a settled signal, the value will be related to the impulse response of the loudspeaker to the frequency spectrum of the center lobe of the tone burst. Thus the frequency resolution of the measurement is reduced, and sharp peaks and dips normally seen in the sine wave response will be obscured. This is illustrated by Fig.18, which shows the frequency response curve obtained using only a single period of the test frequency. This then gives a frequency resolution or bandwidth of the measurement equal to the center frequency of the tone burst. This averaging effect can be seen clearly by comparing Fig.7 with Fig.8 which shows the free-field response measured traditionally with pure sines in an anechoic chamber.

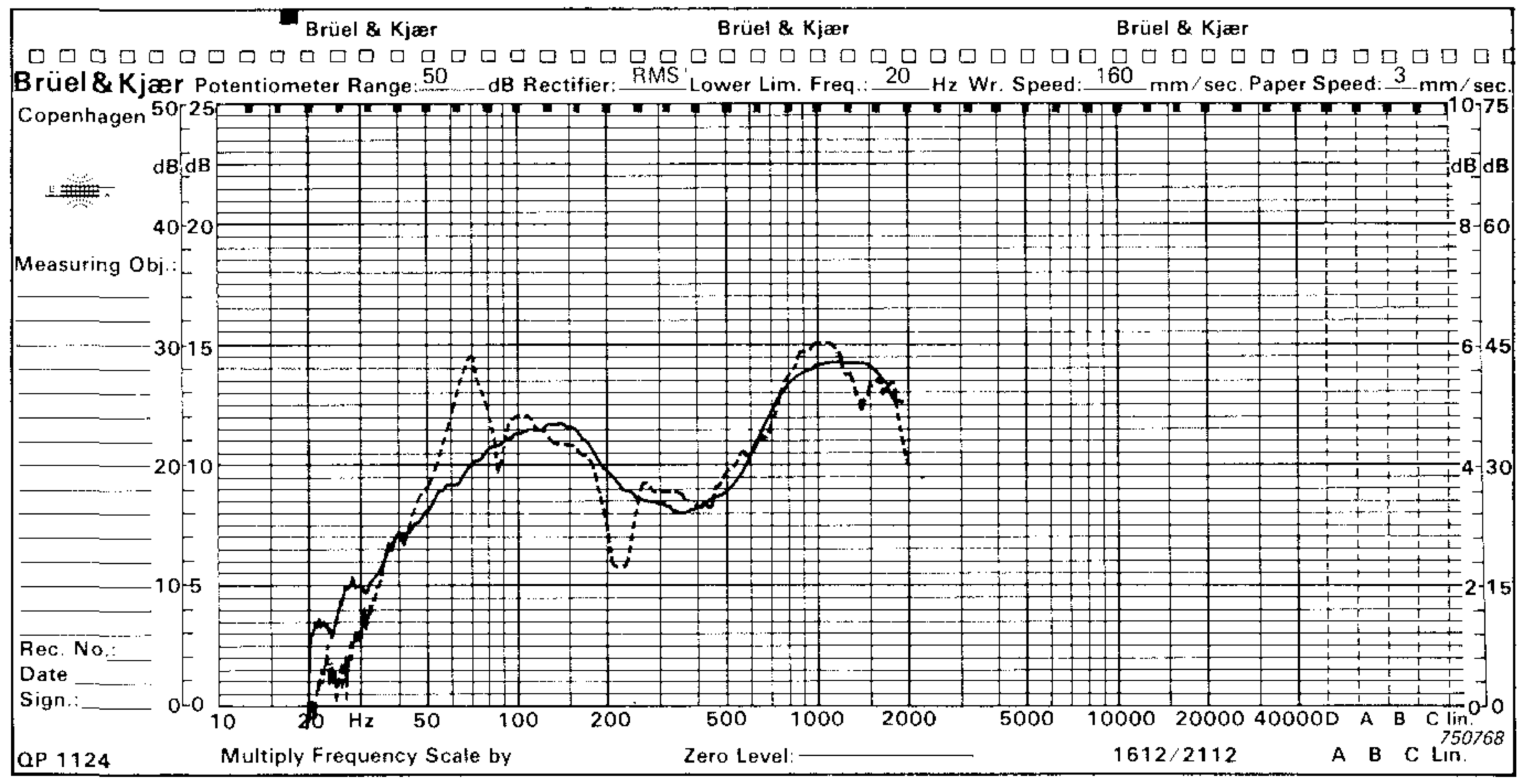

Fig.18. **Illustration of averaging due to too short a tone burst. Dashed line shows free-field sine wave response, solid line shows the response measured with a tone burst consisting only of one period of its frequency**

This does not mean that a measurement is not possible on a single period tone burst. Provided that the loudspeaker is good enough, it can have settled to the correct value in that single period. For this to be possible, the length of the tone burst should be equal to or greater than the settling time of the loudspeaker. Otherwise the averaging phenomenon illustrated in Fig.18 will occur.

In the case of directional characteristics, the broad frequency spectrum of the tone burst will excite resonances and standing waves in the loudspeaker and its cabinet. On axis, the level of these resonances is well below the direct signal and hence they are not seen. However, off axis where the level of the direct signal is significantly lower, the influence of these resonances will be seen.

Evaluation of Anechoic Rooms

The Gating System can be used as a powerful tool in examining an anechoic room for reflective surfaces. A loudspeaker can be aimed at a suspected reflecting surface, such as the construction around the door, and using the Gating System, reflections, if any, are readily observed on the oscilloscope. The measuring gate can then be adjusted to only include the reflection, and the amplitude of the reflection can be plotted as a function of frequency on the level recorder. This same technique can also be applied to determine the absorption coefficient, as a function of angle, of various materials. A significant advantage of this technique is that the materials can be measured on location in the actual mounting.

An example of the influence of a relatively small and seemingly innocent portion of an exposed door frame in an anechoic room is seen in the difference between the directional characteristics in Figs.19 and 20. The use of the Gating System removed the influence of the reflection which was also seen very clearly on the oscilloscope when using a tone burst.

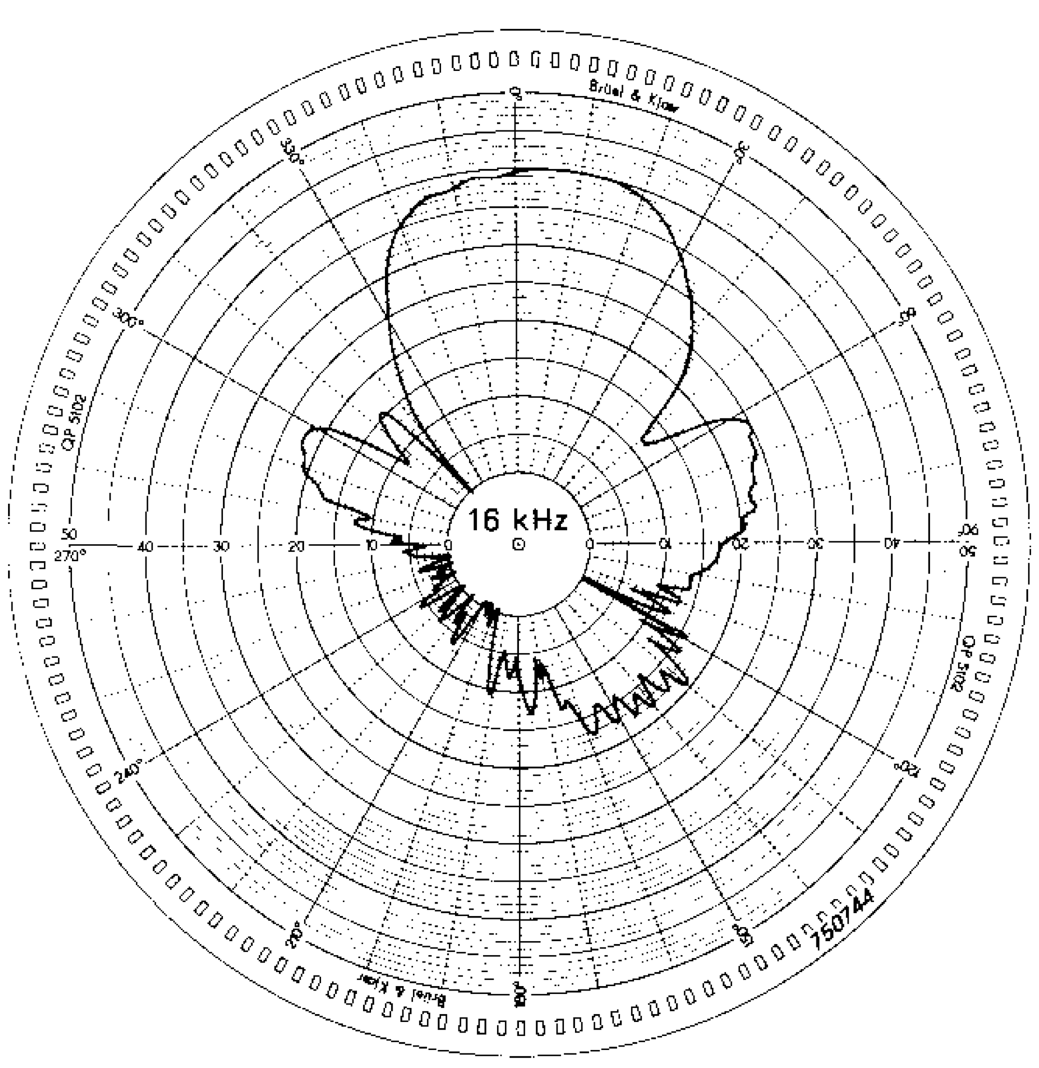

Fig.19. **Directional characteristics of loudspeaker measured using pure sine in anechoic room. Note the influence of reflections from 120° to 170°. These reflections come from the door frame**

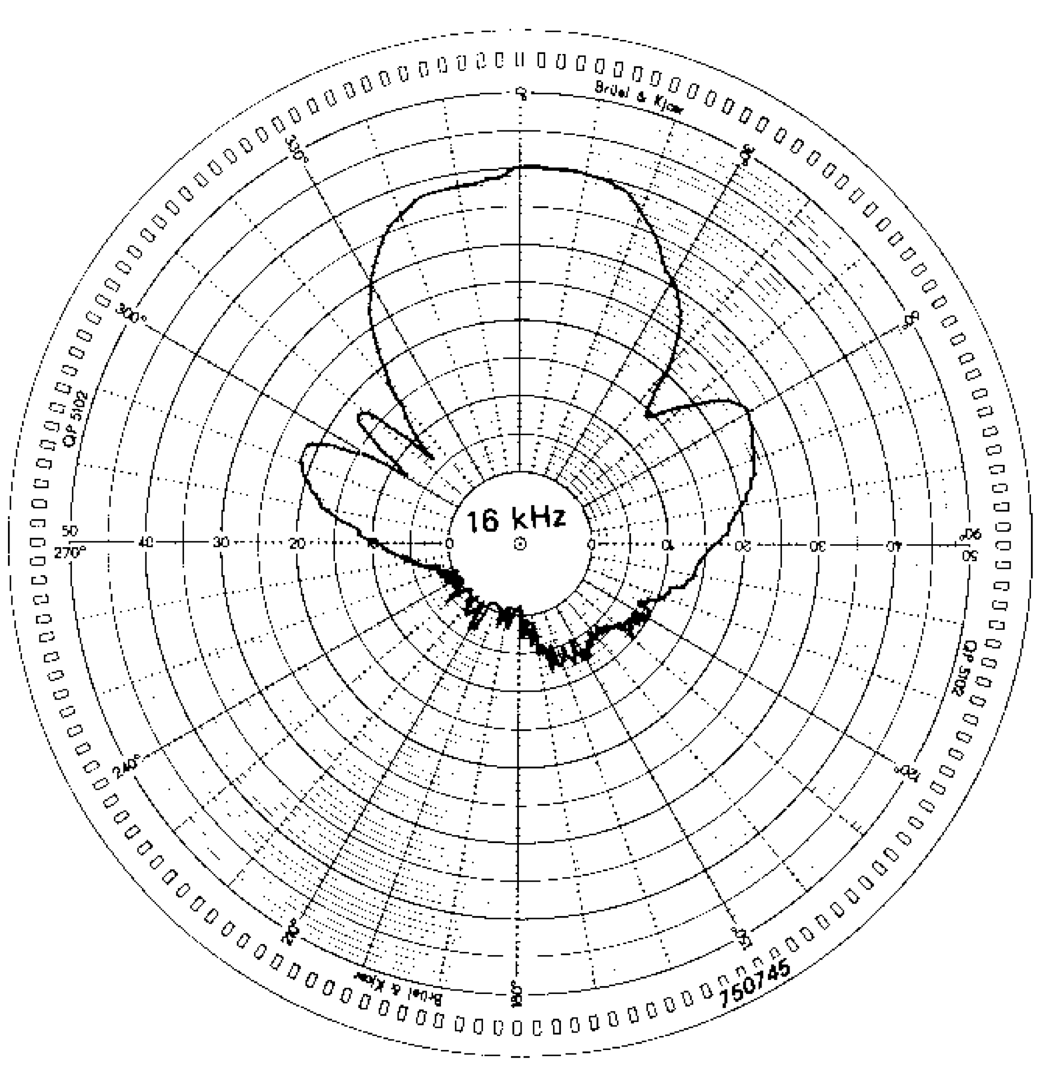

Fig.20. **Reflections from door frame (Fig.19) are removed by use of the Gating System**

Harmonic Distortion

Distortion measurements may be made using a tone burst provided that the response time of the measuring filter is short enough to permit full response. The response time (T) of a filter is related to its bandwidth (B) by

$$T \approx \frac{1}{B}$$

Thus for typical tone bursts of 3 to 10 ms duration, bandwidths of at least 100 Hz to 300 Hz will be required. This requirement is fulfilled by the Heterodyne Analyzer Type 2010 which has bandwidths up to 1000 Hz. Fig.21 illustrates this by showing the response of a filter with a 316 Hz bandwidth to tone bursts of various lengths.

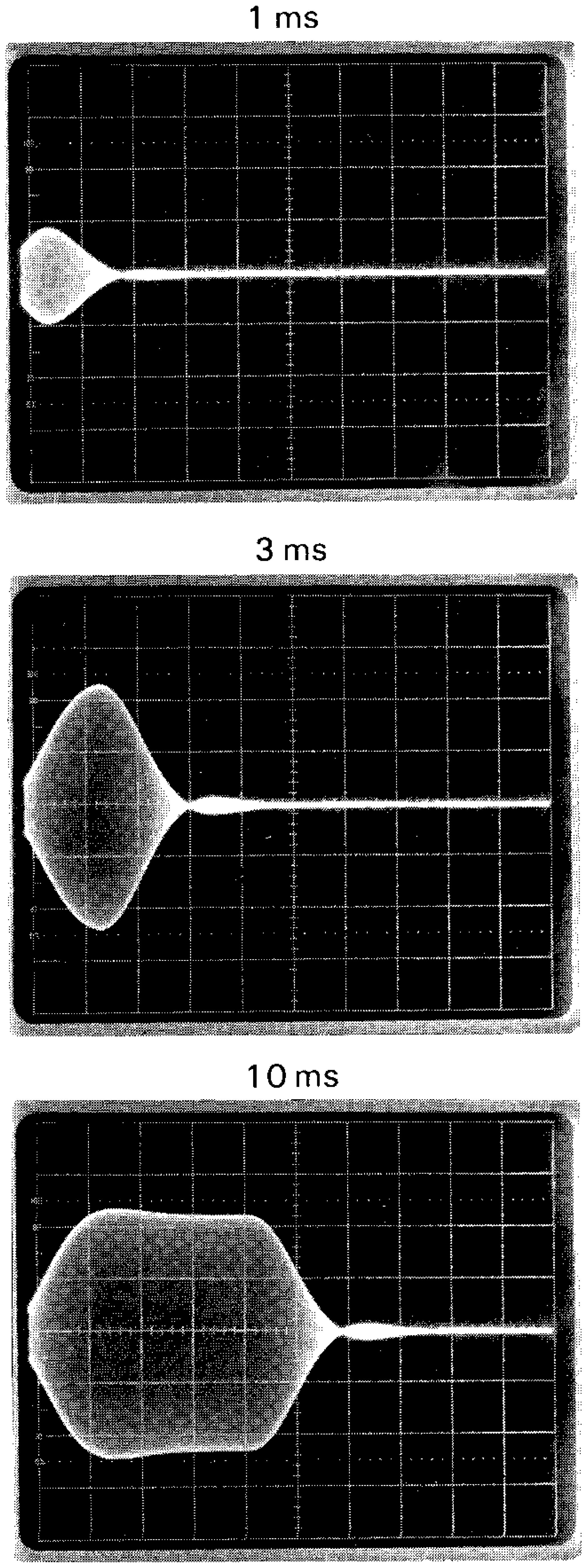

Fig.21. Response of 316 Hz wide filter to various tone burst durations

For distortion measurements, the filter will not be tuned to the fundamental as in Fig.21 but to some harmonic. However, the same rules for filter response time still apply. Fig.22 shows the response of a filter tuned to the second harmonic of the frequency contained in the tone burst. The first peak corresponds to the filter's response to the beginning of the tone burst, and the second peak to the end of the burst. The steady-state portion before the end of the burst corresponds to the second harmonic distortion of the loudspeaker to which the measuring gate is adjusted.

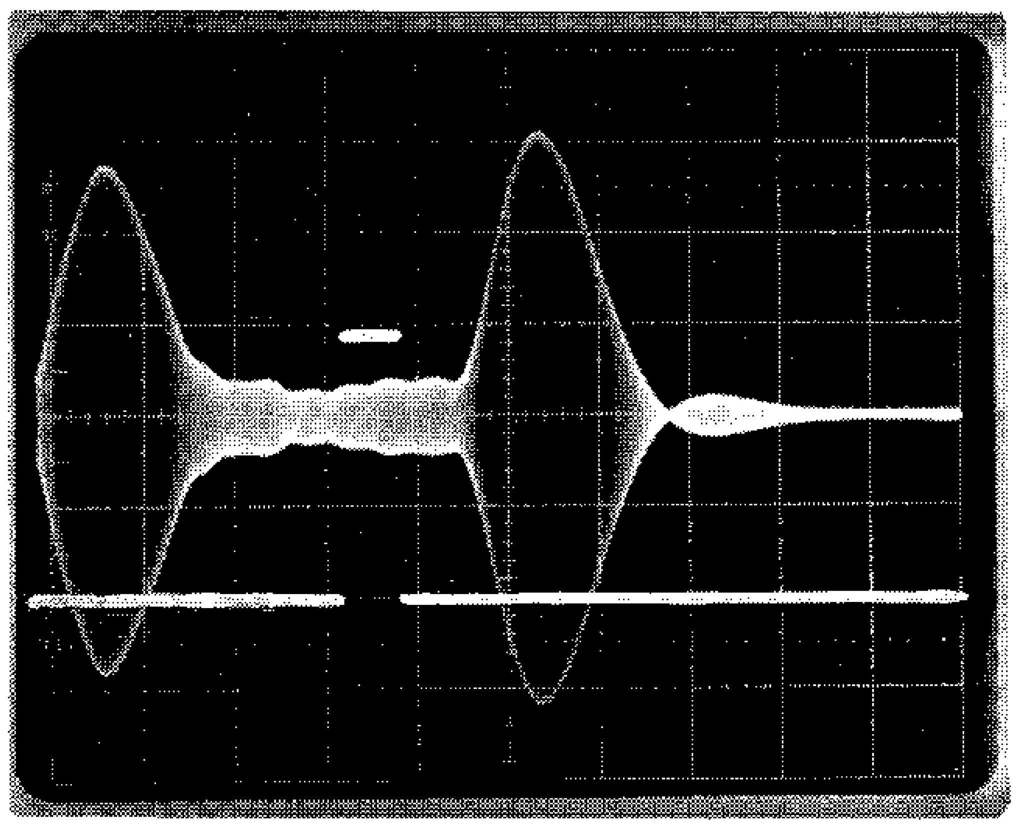

Fig.22. Response of filter tuned to second harmonic of the tone burst. Gate is adjusted to measure distortion component

Fig.23 shows a typical instrument set-up where the Tracking Frequency Multiplier Type 1901 is used to lock on to the chosen harmonic and to tune the analyzer section of the 2010 to that harmonic. (Distortion Measurement Control Unit Type 1902 could be used in place of the 1901.) Typical second and third harmonic distortion curves measured with such a set-up are shown in Figs.24 and 25. Both pure tone and gated measurement curves are shown. It can be seen that due to the wide filter bandwidth (316 Hz) used, the lower frequency limit is in the 500 to 1000 Hz range.

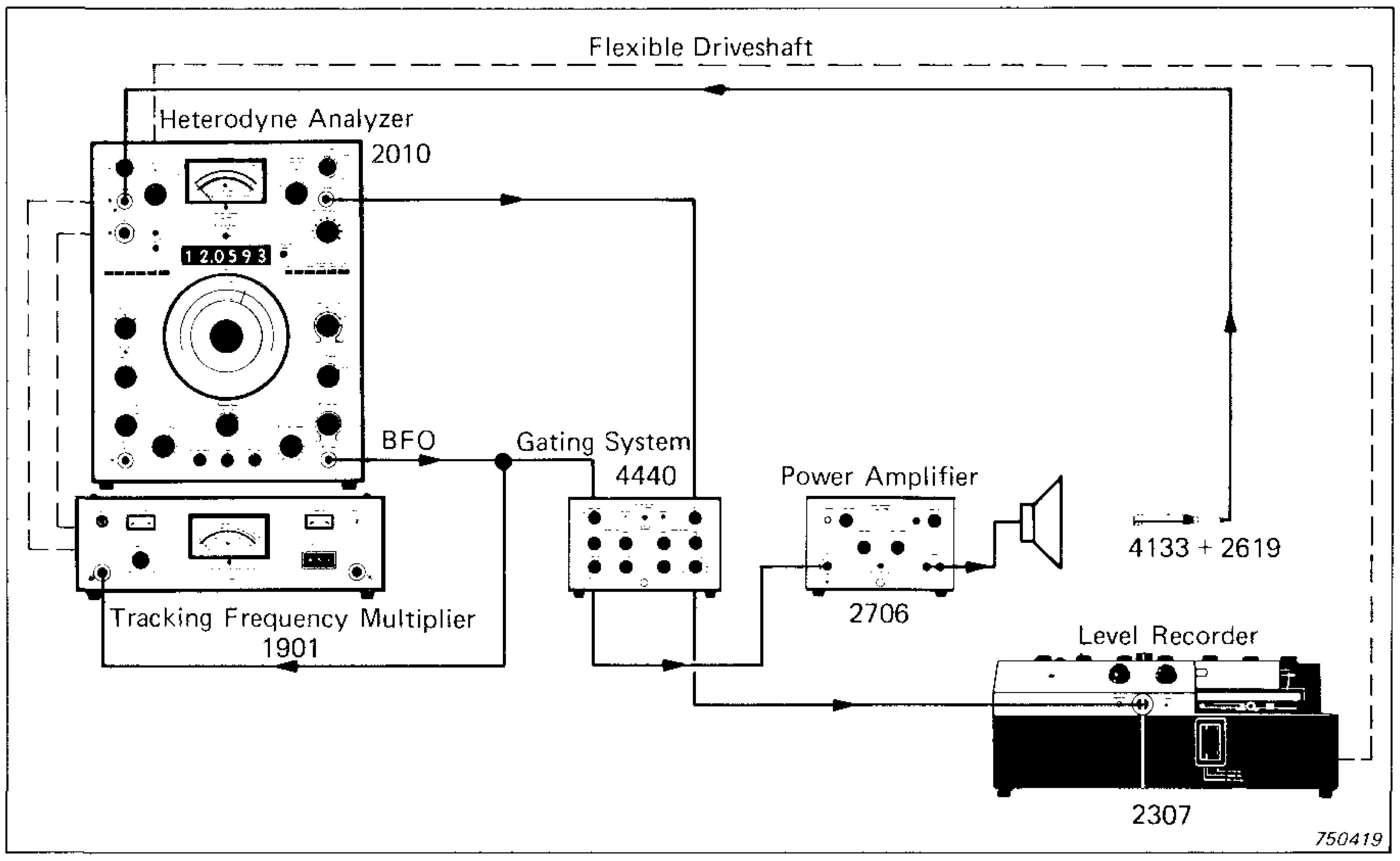

Fig.23. Instrument set-up for distortion measurements using the Gating System

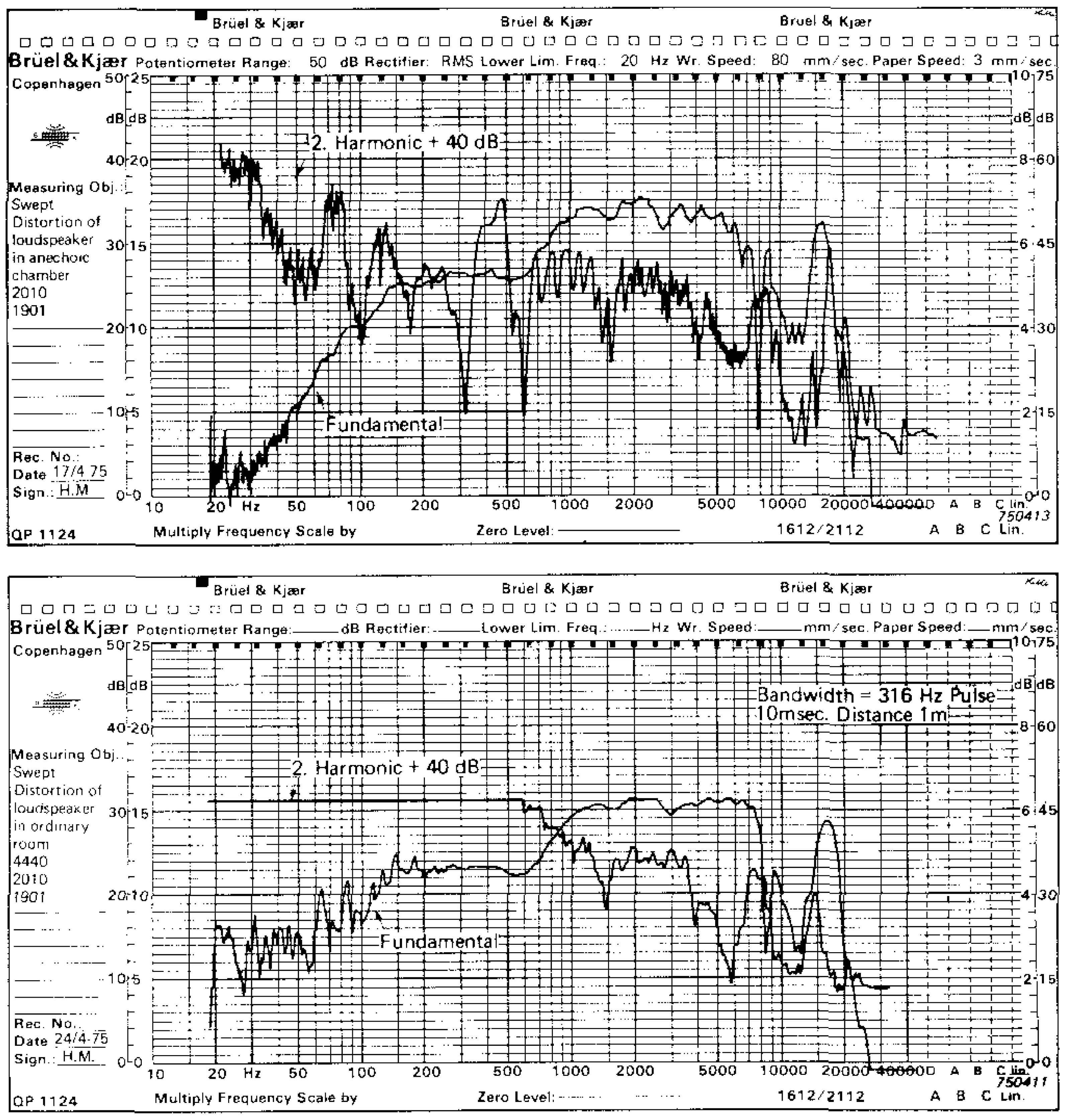

Fig.24. **Results of distortion measurements. The top chart shows the fundamental and second harmonic of a loudspeaker measured with pure sines in a free-field. The lower chart shows the same measurements made with the Gating System in an ordinary room**

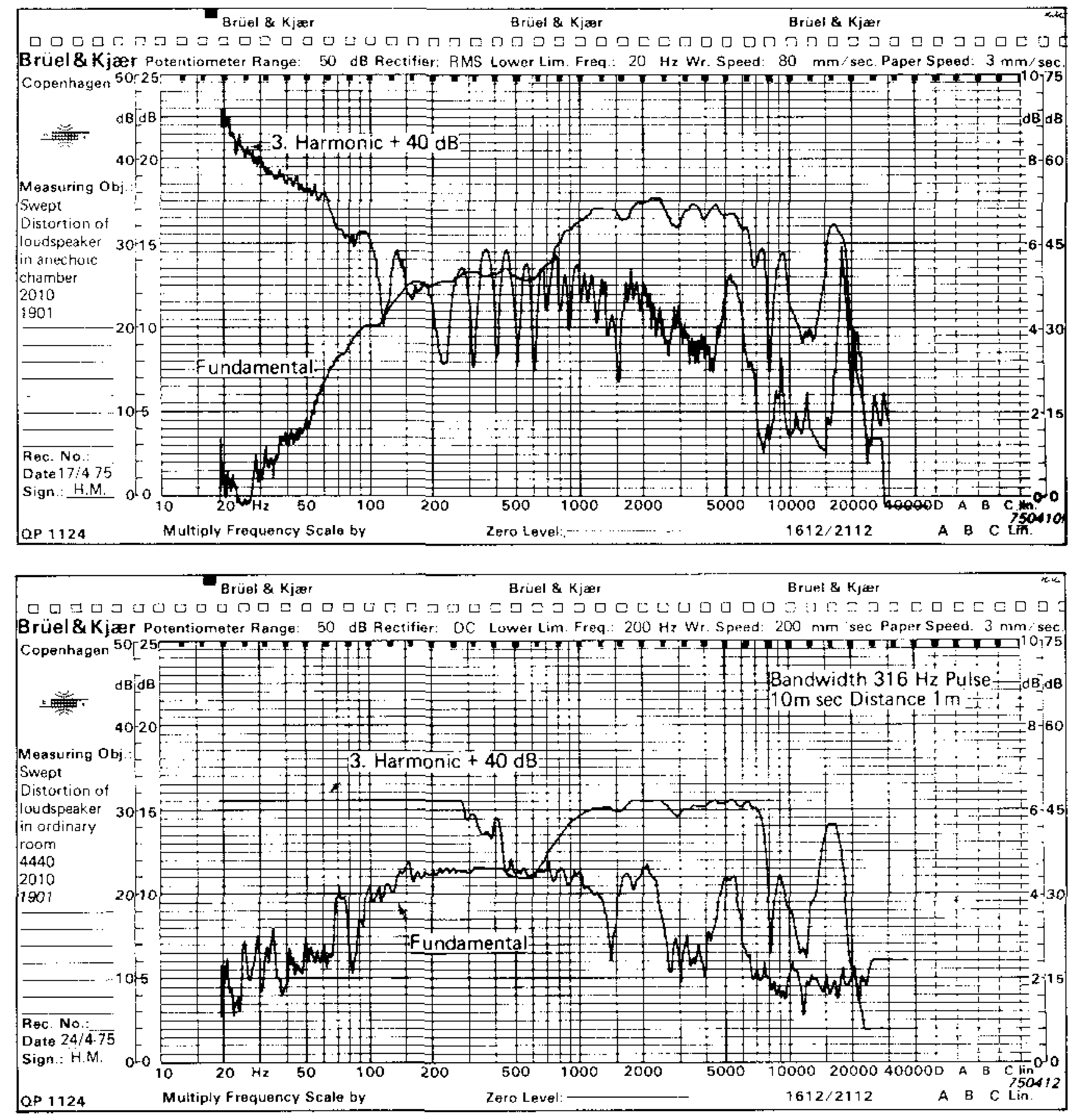

Fig.25. **Results of distortion measurements. The top chart shows the fundamental and third harmonic of a loudspeaker measured with pure sines in a free-field. The lower chart shows the same measurements made with the gating System in an ordinary room**

Tone Burst Response

The Gating System may also be used as a tone burst generator to permit subjective evaluation of loudspeaker transient response as viewed on the oscilloscope. In the use of tone bursts for this technique, it is important to remember the frequency distribution of the energy of the burst as pointed out in section "Tone Burst Characteristics". The duration of the burst should be chosen to give an appropriate bandwidth related to the center frequency.

Early Reflections

The Gating System Type 4440 may also be applied to the measurement of so called "early reflections" which indicates the energy distribution as a function of frequency in the milliseconds immediately after the tone burst was supposed to stop. Early reflections are due to overhang of the speaker, and reflections, standing waves, and resonances in the speaker itself and especially in the cabinet.

A measurement of early reflections indicates how fast the response of the system is and how much sound of its own the loudspeaker box adds to the signal due to internal reflections and resonances. Barlow (Ref.9) and Stevens (Ref.10) have studied the so-called "Box Sound" and have suggested that early reflections may be an explanation of this phenomenon. Thus the lower the level of the early reflections, the better the sound quality.

Since the early reflections are strongly frequency dependent, it may be difficult to find the point to which the measuring gate should be adjusted. But by changing the frequency up and down, the point at which the phenomenon is most pronounced can be observed on the oscilloscope and the gate can be adjusted to measure this. Such an adjustment is shown in Fig.26.

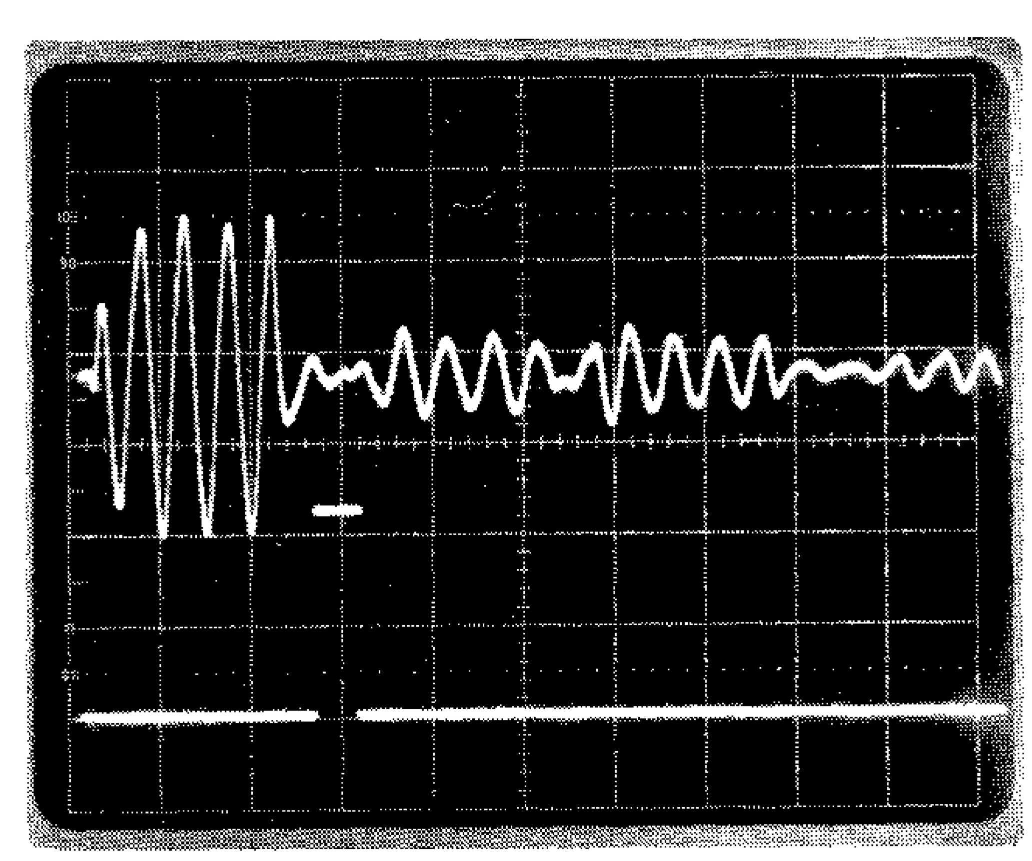

Fig.26. **Adjustment of the gate for measurement of early reflections**

The results of such measurements may sometimes show that a relatively small, and seemingly insignificant peak in the continuous sine wave response curve may be related to a much larger peak in the early reflection curve. Thus the coloration of the sound at that frequency will subjectively be much worse than indicated by the sine wave resnonse. (Fig.27).

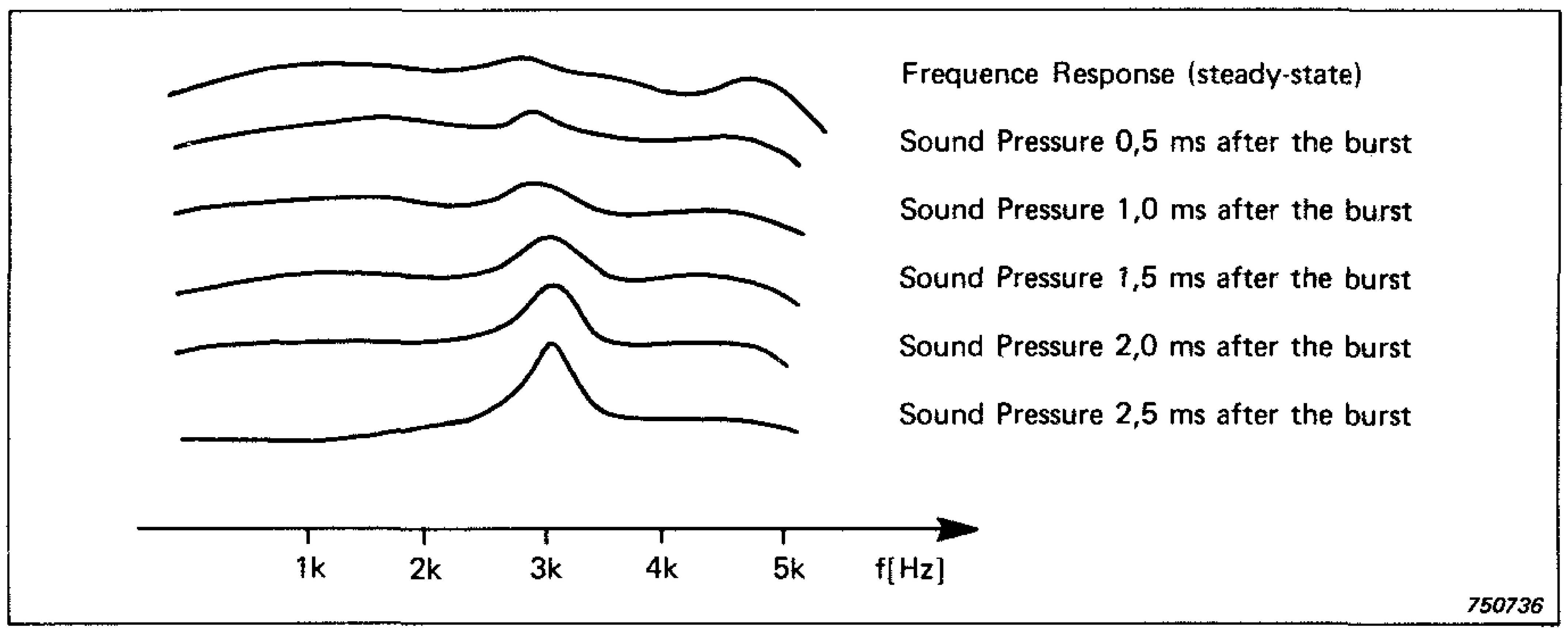

Fig.27. Frequency characteristics of early reflections compared to the steady state response

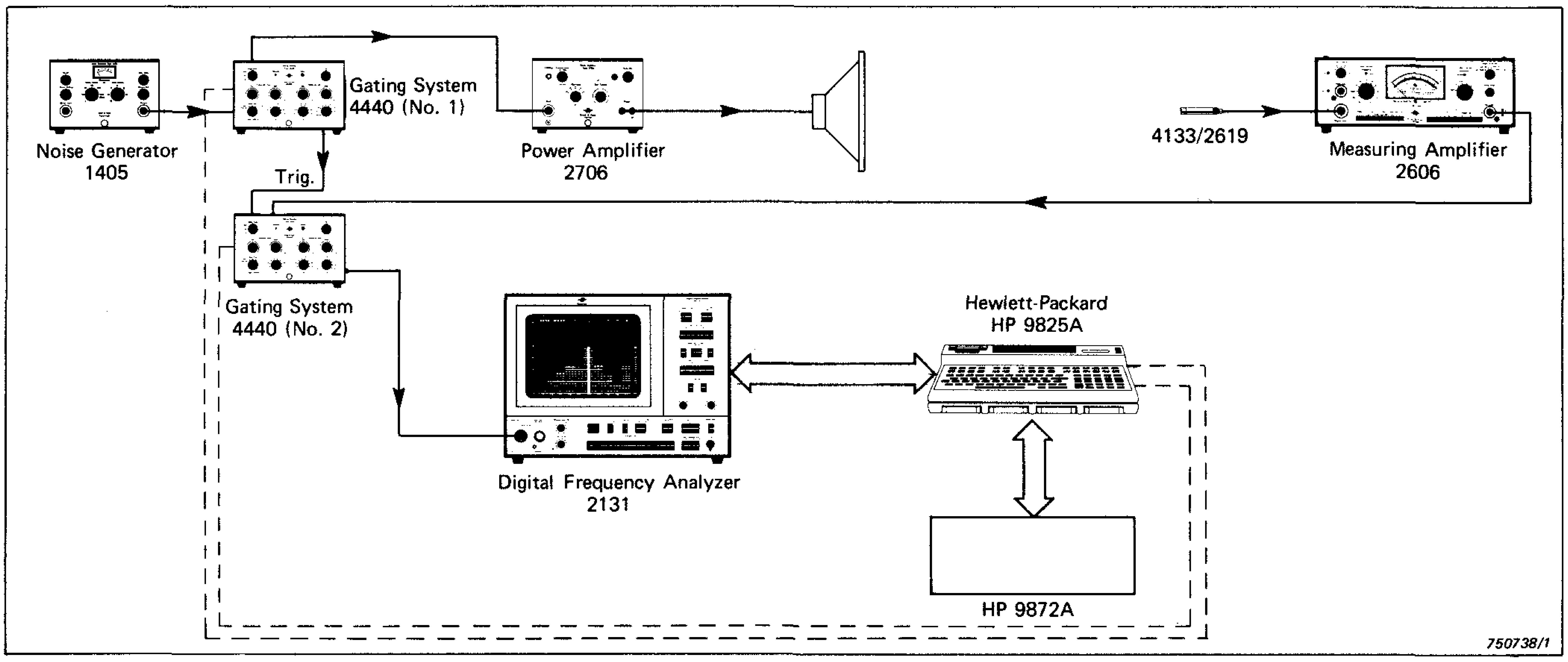

Fig.28. Instrument set-up for real-time analysis of early reflections using gated white noise

A considerably more advanced technique for studying early reflections is suggested in Fig.28. Here, the white noise output of Noise Generator Type 1405 is gated and fed to the speaker. (It should be noted that gated white noise retains its white noise spectrum). A second Gating System is used to process the received signal. The signal is fed through the transmitting section of the second 4440 which is triggered from the Gate Output of the measuring section of the first 4440 (Fig.29). The output of the second 4440 (Pulse Output) is then fed to the Real-Time Analyzer Type 3348 where the frequency spectrum of the desired section of the early reflection is desplayed. The delay of the gate can then be varied and the changes in the frequency spectrum can be observed as a function of time after the end of the burst.

This same instrumentation can also be used to analyze the characteristics of loudspeakers which are designed to use reflections from walls or corners to give the desired sound. Such types include omni-directional loudspeakers and speakers designed to be placed a given distance from a wall to use its reflections. By adjusting the gate to include the last part of the tone burst and the first several reflections, the response of this speaker type and the room's early reflections can be seen.

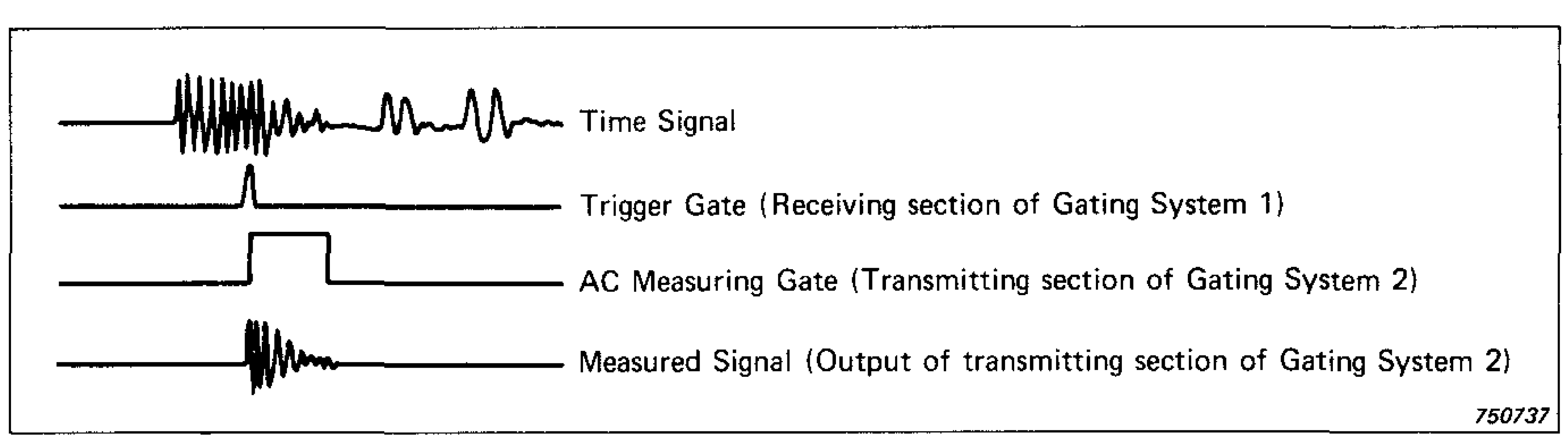

Fig.29. Setting of gates to measure early reflections with the instruments in Fig.28

Phase Response

A tone burst test signal not only contains the amplitude information, but also the phase information. This can be measured using a phase meter with a sufficiently fast response time.

The instrumentation and techniques described in B & K Application Note No. 15-090 "Loudspeaker phase measurements, transient response and audible quality" can also be used for phase measurements using tone bursts. However, to make this possible, two minor modifications are required, one to increase the phase meter's response time, and the second to DC couple the receiving section of the Gating System. For information on this, contact Brüel & Kjær directly.

The system has the same basic low frequency limits as for amplitude response measurements ... being dependent, of course, on room size.

Conclusion

Gating techniques provide a powerful tool extending electro-acoustic measurements to non-anechoic rooms. By the use of the Brüel & Kjær Gating System Type 4440, amplitude and phase response, distortion and directional characteristics may be measured under simulated free-field conditions.

The transmitted tone burst contains a broad spectrum of frequencies. However, by selecting only the steady-state portion of the burst using a suitably delayed measuring window, the single tone response is measured, without the influence of the broad band spectrum. By adjusting the gate to other sections of the burst, the transient response and early reflection characteristics of the loudspeaker can be measured.

Appendix

Calculation of Microphone/ Loudspeaker distance and pulse length

Assume a room (Fig. 30) with the transducers equally spaced between floor and ceiling (h, the height of the room is assumed the smallest of the room's dimensions). First, we will only consider reflections from side walls, ceiling and floor. The pulse length (t) must then be shorter than the difference between the time it takes to travel the reflected (2l/c) and the direct path (d/c). Hence

$$t \leqslant \frac{2l - d}{c} = \frac{\sqrt{h^2 + d^2} - d}{c} \qquad (1)$$

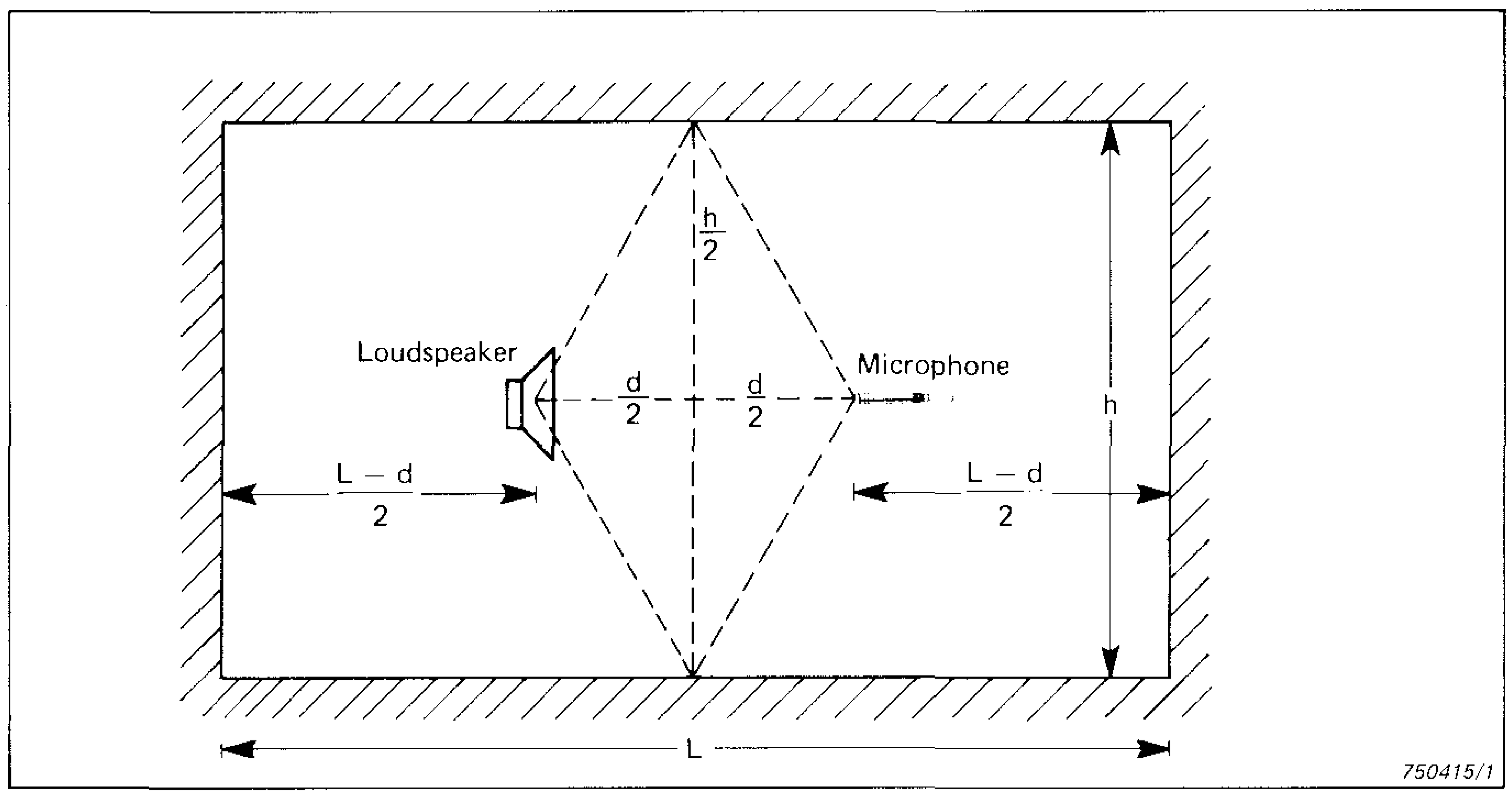

Fig. 30. Travel distances for the first reflections when the loudspeaker and microphone are centered along all three axes of the room

Solving for d we obtain

$$d \leqslant \frac{h^2 - c^2 t^2}{2\,ct} \qquad (2)$$

The criterion that the microphone should be at least one wavelength from the loudspeaker gives

$$d \geqslant ct \qquad (3)$$

where t is the period at the lowest frequency which also corresponds to the pulse length which contains one period at the lowest frequency. Setting Equations 2 and 3 equal we obtain the optimum pulse length and corresponding transducer spacing:

$$ct = \frac{h^2 - c^2 t^2}{2\,ct}$$

$$3\,c^2 t^2 = h^2$$

$$t = \frac{h}{c}\frac{\sqrt{3}}{3} \qquad (4)$$

$$t = \frac{h}{595} \qquad (5)$$

The reciprocal of which gives the lower frequency limit $f_{min.}$

$$f_{min.} = \frac{595}{h} \qquad (6)$$

at a distance between transducers of

$$d = ct = c\left|\frac{h}{c}\frac{\sqrt{3}}{3}\right| = 0{,}577\,h \qquad (7)$$

which is the optimum spacing between transducers for a given minimum room dimension h.

For reflections from the end walls of the room along its longest dimension (L), the length of the pulse must be shorter than the difference between the time it takes for the first reflection to return to the microphone (L/c) and the time it takes for the direct sound to reach the microphone (d/c).

Hence:

$$t < \frac{L - d}{c}$$

or

$$d < L - ct \qquad (8)$$

Now reflections from the far wall only become a limitation when the minimum distance of Equation (8) is equal to, or less than that of Equation (3). Setting the two equal

$$L = 2\,ct \qquad (9)$$

and substituting t from Equation (4) we get

$$L = \frac{2}{3}\sqrt{3}\,h = 1{,}15\,h \qquad (10)$$

Hence the length of the room must be at least 15% longer than the smallest dimension in order for Equations 5-7 to be valid.

However, with reflections from the end walls setting the limits, the pulse length must be (from Eqn. (9))

$$t = \frac{L}{2c} = \frac{L}{688} \qquad (11)$$

with an optimum distance between transducers of (combining Eqn. (3) and (9))

$$d = \frac{L}{2}$$

Sweep Speed Derivation

For the logarithmic Level Recorder paper commonly used with Brüel & Kjær generators, one decade in frequency corresponds to 50 mm paper displacement.

Hence the paper displacement (x) is related to frequency (f) by

$$x = 50 \log f$$

differentiating with respect to time

$$\dot{x} = \frac{50 \log e}{f}\,\dot{f}$$

$$\dot{x} = \frac{21{,}7}{f}\,\dot{f}$$

where $\dot{x}$ is the Paper Speed (P) in mm/s and $\dot{f}$ is the sweep rate in Hz/s. Hence

$$\dot{f} = \frac{Pf}{21{,}7} \qquad (12)$$

With the Gating System, the frequency resolution (B) in Hz equals the sweep rate divided by the number of measurements per second (the Repetition Rate, R).

$$B = \frac{\dot{f}}{R}$$

$$BR = \dot{f} \qquad (13)$$

Combining Equations 12 and 13 gives

$$BR = \frac{Pf}{21{,}7}$$

or

$$P = \frac{21{,}7}{f}\,BR \qquad (14)$$

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